

Amadeus Pro

The swiss army knife of sound editing



User reference guide for v1.4

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1 Introduction

Thank you very much for choosing to use **AmadeusPro**. This is a very high quality feature-packed audio editor which is nevertheless easy to use and very reasonably priced.

This manual will guide you through the main features of **AmadeusPro**. It is not intended to be an exhaustive reference text, but it will focus on those features that are maybe not absolutely intuitive at first glance. The best way by far to find out how a feature that is not covered in detail by this manual works is to simply try it out and see what happens. Most of the time, it should be pretty much what you expected from the name of the command and / or from the conventions in use by most Mac OS applications. You should not be shy about trying functions out, since destructive actions can always be undone, unless a specific warning is given that this is not the case.

This manual also assumes that you are reasonably familiar with computers and know how to use basic Mac OS applications. (Think of this as your car manufacturer assuming that you actually passed your drivers license test before buying a car from him.) In particular, it assumes that you know about files, windows, the menu bar, etc. If you have little familiarity with sound editing, it is probably a good idea to start by reading Sections 2 and 5 in their entirety, since these sections explain the basic terminology and concepts used in this manual.

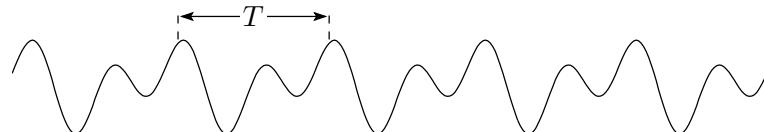
This manual is a PDF file with hyperlinks. All words written in blue (this includes not only URL's but also references to sections and page numbers for example) are clickable. Some terms that may not familiar to everyone are linked to their [Wikipedia](#) entries. This is why no glossary is provided, since the Wikipedia entries are usually quite well written and more complete than any glossary that could be provided here. It is hoped that these Wikipedia links are stable, but it may of course happen that they break sometime in the future. If you find a broken link, please inform me by sending an email to Martin@HairerSoft.com.

2 Basic notions of sound editing

2.1 What is a sound?

Let's start with the basics since, in order to understand how a sound is represented and can be manipulated by a computer, one should have some understanding of what a sound actually is physically. In a nutshell, a sound is a small rapid variation in the air pressure. The variation of pressure is captured by your internal ear and transmitted to the brain, which interprets this signal as a sound.

A microphone is nothing but an artificial ear. It measures the air pressure and transforms it into an electric impulse. A sound is therefore usually represented as a function representing the air pressure (or equivalently the voltage at the exit of the microphone) as a function of time. A typical representation of a sound signal would be:

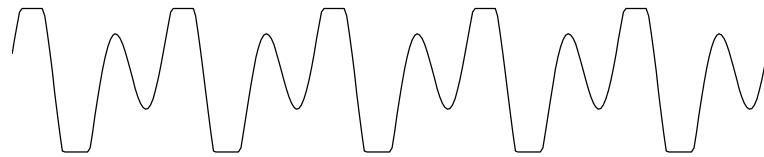


Note that, very often, the signal is (at least approximately) periodic as in the picture above. The time interval between two identical looking chunks of the signal, denoted by T in the picture above, is called the *period*

of the signal and is usually measured in milliseconds. The inverse of the period is called the *frequency* of the signal and is measured in Hertz, also abbreviated by Hz. Some high-frequency sound signals are measured in kiloHertz (that is thousands of Hertz), also abbreviated by kHz. A Hertz is the inverse of a second. So a sound with a period of 0.05 seconds has a frequency of $1/0.05 = 20$ Hertz. The human ear can hear sounds that have frequencies ranging approximately between 20Hz and 20kHz. These bounds however vary with age, with older people less able to perceive high frequencies. Sounds with frequencies lower than that are called infrasounds and sounds with frequencies higher than that are called ultrasounds.

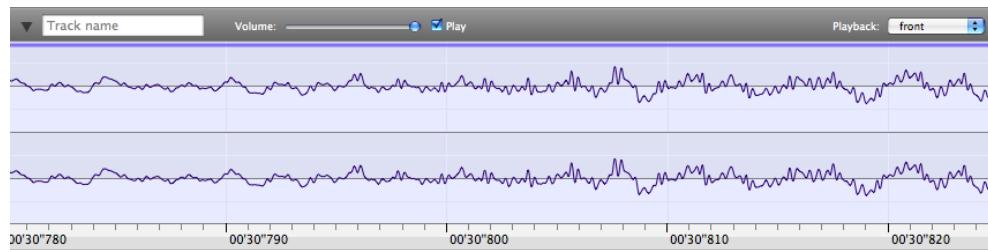
2.2 How does a computer represent it?

A computer can only handle discrete pieces of data. A signal like the one depicted above will therefore be transformed into a set of discrete values. Each of these values is called a *sample* and the number of values per second used to represent the signal is called its *sampling rate*. Therefore, if a signal has a sampling rate of 44.1kHz (the rate used in audio CDs), then the signal is measured 44100 times per second. The individual values making up the signal are of course discrete as well. If the signal is stored in the computer by using 2^N different values, it is said to have a *sampling depth* of N . Sound on an audio CD has a sampling depth of 16, meaning that it is stored using $2^{16} = 65,536$ different values. The characteristics (sampling rate and sampling depth) of a sound are displayed in the bottom left corner of the sound window. For practical purposes, it is usually assumed that samples take values between -1 and 1 . In the previous example of a 16bit sound, this means that every sample can take 65,536 values that are equally spaced through the interval $[-1, 1]$. If a sound containing samples that are close to these limits is amplified, a phenomenon known as ‘clipping’ occurs. This is when, as a result of a sound effect being applied to it, some samples ‘would like’ to take values outside of the $[-1, 1]$ range and are forced back into it by the computer. If the sound signal represented above was to be amplified too much, it could therefore end up looking like this:



When clipping is severe, it introduces very noticeable artifacts that are quite difficult to get rid of, one should therefore take care to avoid it.

Humans perceive sounds with two ears. If one wishes to reconstruct accurately an auditory experience, it is common to use two microphones, one for each ear. This is called a *stereo* sound. When only one microphone is used, it is called a *mono* sound. The following screenshot shows how a stereo sound is displayed in **AmadeusPro**:



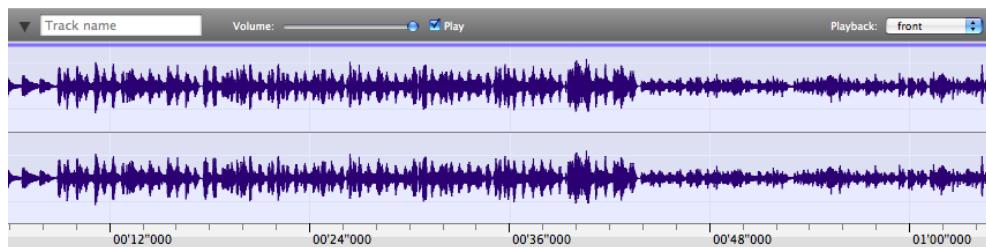
The upper signal corresponds to the left ear and the bottom signal corresponds to the right ear.

Note that the structure of the signal (also commonly referred to as the ‘waveform’) only becomes apparent when one zooms into the signal. For example, in the example above, the width of the whole display only corresponds to about 50 milliseconds (one twentieth of a second). Zooming into the display can be achieved with the **Horizontal zoom** slider:



The number appearing next to the slider when you move it represents the number of samples of sound that fit into one pixel of your screen.

When a larger portion of the sound is shown, a typical display would rather look like this:



Here, you can see that the width of the display is about one minute. At this scale, the structure of the waveform is not apparent anymore. What you can see essentially represents the volume of the sound as a function of time (the fatter the dark blue area, the louder the sound).

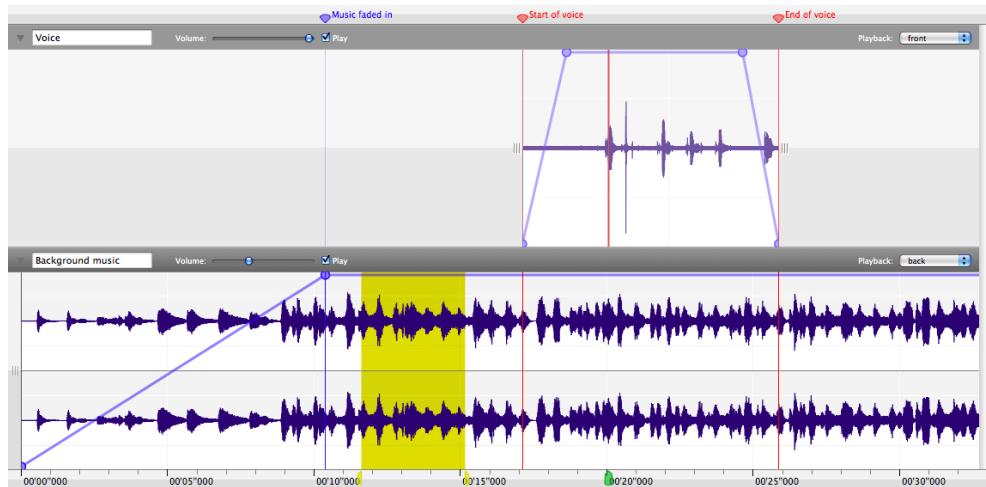
2.3 Sound intensities: dB versus percentages

The intensity of a pure sine wave is defined as the square of its amplitude. It is normalised in such a way that a pure sine wave with the maximal amplitude has an intensity of exactly 1. **Decibels** (abbreviated as dB) are a logarithmic unit that better reflects the fact that our perception of sound intensity is not linear. A *difference* of 10dB corresponds to a *ratio* of 10 in intensity. It is normalised in such a way that a sine wave with maximal intensity 1 corresponds to 0dB and a sine wave with intensity 0.1 corresponds to -10dB . This is why intensity values displayed in dB will always be *negative*.

For ‘physical’ rather than ‘digital’ sounds, the reference is usually chosen such that 0dB corresponds roughly to the threshold of human hearing, so you will often see sound intensity measurements with *positive* dB values. Since a given physical sound can end up at any intensity in the computer, depending on the setting of both the recording and the reproduction equipment, there is no convenient way of converting between the two. However, *differences* in dB values have the same meaning in both cases.

2.4 Anatomy of an AmadeusPro document

An **AmadeusPro** document consists of one or more tracks. Each track consists of either a mono or a stereo sound. Normal sound documents stored in Mp3, WAVE or AIFF files for example typically consist of only one track. A typical document with several tracks will be displayed by **AmadeusPro** in the following way:



The next sections are a short guided tour through the various elements that can be seen in this picture.

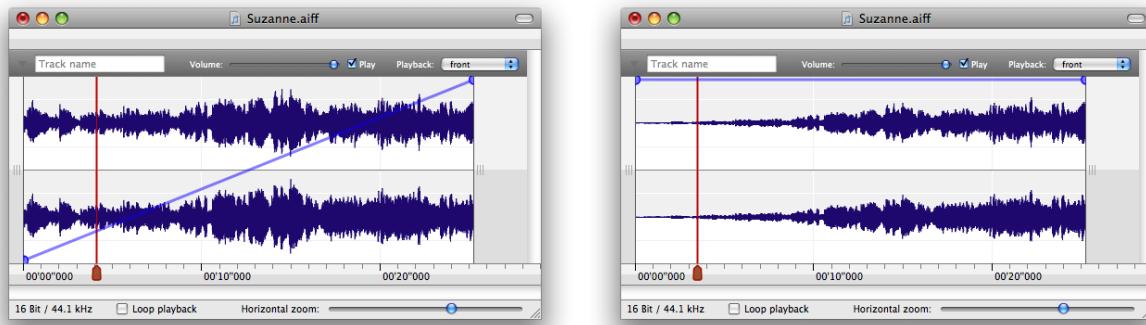
2.4.1 Tracks

This document consists of two tracks. The first track, called ‘Voice’ (tracks can be given arbitrary names) is a mono track. On a multichannel sound interface, it would be played back in the front. The second track, called ‘Background music’ is a stereo track. On a multichannel sound interface, it would be played back in the back. For more information on sound playback, see Section 4.2. In particular, see Section 4.3 for information about the meaning of the **Playback** pop-up menu. One of the tracks is considered to be the current track. The other tracks are drawn in a slightly lighter shade (in this example, the current track is the ‘Background music’ track). The current track can be changed by simply clicking anywhere in it. This is the track on which sound effects and editing operations are applied.

2.4.2 Playback volume and envelope

You can also see on this picture that each track can be associated with a playback volume. In this example, the ‘Background music’ would be played at a lower volume than the ‘Voice’ track. Furthermore, each track has a playback envelope, that is represented as a thick blue line which connects knobs. New knobs can be added to the playback envelope by simply clicking on it. A knob can be removed by dragging it out of the window. The first and the last knob are fixed and cannot be removed. The playback envelope allows you to change the volumes of different sections of a sound independently.

The **Apply Envelope to Track** function from the **Sound** menu allows you to apply the playback envelope to the current track. Here is an example illustrating this:



In the first picture, you see a track with a volume envelope that causes the volume of the track to gradually increase from 0 to the maximal volume. After selecting **Apply Envelope to Track**, this changing volume has been applied to the waveform, which you can now see taking increasingly larger values as time goes on. The playback envelope on the other hand has been reset to the default value of being constant at the maximal value throughout the whole track. Playing back the track produces the exact same output in both cases.

2.4.3 Selection

Each of the tracks can have a portion of it selected. In this example, the portion of sound between about 11.5 and 15 seconds from the start is selected in the ‘Background music’ track. The ‘Voice’ track has an empty selection. In this case, an insertion point is drawn in red. This is where the program would insert a sound if you copied and pasted it from another document. The selection can be changed by clicking into the sound and then dragging the mouse (with its button still pressed down) until the desired portion is selected.

If a track contains two channels, it is possible to select only the left (respectively right) channel by clicking near the top (respectively the bottom) of the track in question. In this case, the ‘cross’ cursor will be modified to show a small ‘L’ (respectively ‘R’). This allows for example to apply an effect to one channel only.

It is possible to preview the effect of an edit before actually performing it. If the selection is empty and one hits the ‘e’ key, **AmadeusPro** will play a short portion around the selected area of the current track, but *omitting* the selection in question. This allows to check for example whether cutting away a piece of sound will introduce a small artifact or not. One way of avoiding the creation of artifacts is to use the ‘Extend to Crossing’ function in the ‘Selection’ menu. The length of the preview can be adjusted with the ‘Length of edit preview’ function of the ‘General’ preferences pane.

2.4.4 Playback head

The green shape located in the timescale at around 20 seconds is the playback head. This is where playback would start (resume) if you hit the space bar. For more information on sound playback, see Section 4.2.

2.4.5 Markers

As one can see in the screenshot, a document can have an arbitrary number of markers ('Music faded in', 'Start of voice', etc). These are just handy ways of remembering a particular location in a document. In order to create a marker, simply click in the white / gray area at the top of the window or move the insertion point to the correct location and type 'm'. Markers can also be given colors, for example in order to distinguish between markers that serve different purposes. One convenient way of working with markers is the **Markers** tool that can be accessed from the **Windows** menu. When you click on a marker in this tool, the display of the current window changes so that the clicked marker is moved to the center of the window. If you use shift-click to select several markers, the display is zoomed out (or in) in such a way that all the selected markers fit within the width of the current window. If **Adjust selection** is checked, then the selection of the current track is adjusted so that it selects the portion of sound between the chosen markers. If only one marker is chosen, it moves the insertion point to the location of that marker. The '+' button creates a marker at the center of the selection of the current track or at the insertion point if the selection is void. The '-' button allows you to delete the selected marker(s).

Markers:	
Name	Location
Music faded in	02'01"961
Start of voice	02'25"251
End of voice	02'44"361

+ - Adjust selection

The four last items in the **Selection** menu are devoted to the management of markers. Since the meaning of the first two should be obvious, let us only concentrate on the function of the remaining two:

- **Generate Markers...** Allows to generate equidistant markers on your sound. This is especially useful in conjunction with the **Split According to Markers...** function if you want to cut a very long sound into pieces of reasonable size.

You can choose **Search for silences** in order to place markers at the location of silences in your sound. The **Maximal level** parameter gives the threshold under which a sound is considered to be silence. The **Minimal length** parameter gives the minimal length a silence must have in order to be taken into account.

If a number is found in the marker name entered in the **First marker** text field, it is automatically incremented for each subsequent marker.

- **Modify Markers...** This item allows to modify all the markers of a file in one go. If **Change name** is checked, **AmadeusPro** will use the corresponding name for the first marker and will try to make a guess on how to increment it for the subsequent markers. For example, if you specify 'Track 01', the next markers will be called 'Track 02', 'Track 03', etc. The **Align position** field allows to align the marker positions on a specified time grid. For example, specifying '75 Hz' for a file sampled at 44.1kHz allows to align markers with sector boundaries on an Audio CD (tracks on Audio CD's always have a length that is a multiple of 1/75th of a second). This is useful to burn a CD without gaps between songs.

2.4.6 Metadata

An **AmadeusPro** document can furthermore contain arbitrary amounts of metadata, that is data which is not directly relevant to the editing and playback of a sound, but which gives some additional information. Examples of metadata are the song and artist's name, album cover artwork, etc.



Metadata can be viewed and edited by clicking on the **Information** button in the toolbar or by selecting **Show Sound Info...** in the **Sound** menu. This will also show some generic information about the sound, like its length, file format, sampling rate, etc.

3 File Formats

AmadeusPro supports a variety of popular sound formats, as well as its own ‘Amadeus’ format. Most formats have some limitations that prevent them from being able to represent an **AmadeusPro** document faithfully. It is therefore recommended to always work in the ‘Amadeus’ format and to export the document to a standard format only in the very last step. The shortcomings of the various formats will be explained below.

As a general fact, sound documents can take up a large amount of disk space, and so many formats include some compression algorithm aimed at minimising the size of the resulting file. There are two families of such compression algorithms: *lossy* algorithms and *lossless* algorithms. Lossy algorithms allow for high compression rates, but this comes at the cost of a loss of information. In other words, every time you decode a document compressed with a lossy algorithm (for example to edit it) and then save it back, the sound will lose a bit of its quality. Lossless algorithms on the other hand preserve the data in its entirety, so that there is no loss of quality coming from successive edits. The downside of course is that lossless algorithms do not usually achieve very high compression rates.

If you wish to edit a file encoded with a lossy algorithm, it is therefore recommended to save intermediate steps of the editing process to an uncompressed format or a format using a lossless algorithm.

Typical lossy algorithms are Mp3, AAC and Ogg Vorbis. The AIFF and WAVE formats store uncompressed data by default but also allow you to use lossy compression algorithms like μ -Law, a-Law, MACE-3, MACE-6, QDesign Music 2, and AMR Narrowband. It is beyond the scope of this manual to discuss the suitability of these algorithms for various applications. (Please don’t send me emails asking for advice. Try to experiment by yourself or look for advice on the internet, there is plenty of information available online.) Typical lossless algorithms are FLAC and Apple Lossless.

Some algorithms compress data by a fixed factor (μ -Law and a-Law compress by a factor 2, MACE-3 and MACE-6 compress by factors 3 and 6 respectively), but most other algorithms support several compression factors. A standard measure of compression is the *bitrate* which measures the amount of diskspace (in bits per second) which the resulting file takes. A bitrate of 128kbps corresponds to about 16kB per second, which is roughly 10 times less than the amount of disk space taken by uncompressed sound in CD quality (44.1kHz, 16Bits, stereo). Smaller bitrates of course come at the expense of lower sound quality.

Some formats are not suitable to retain all the attributes of an **AmadeusPro** document. If this is the case, a warning will be shown, together with a short explanation of the features that may be lost. The only way to be 100% certain that all features are retained is to save a document in the **AmadeusPro** format.

The different file formats have specific settings that can be accessed by clicking on the **Settings** button in the file saving dialog. Below is a short description of each of them.

3.1 The Amadeus format

This format is a faithful snapshot of the state of an **AmadeusPro** document. It is the only format that supports all of **AmadeusPro**’s features without any restriction. It can only contain uncompressed data and

has no particular settings.

3.2 The Wave format

This format is very popular mainly on the Windows platform but also on other operating systems. It usually contains uncompressed data, but also supports a variety of compressed formats. **AmadeusPro** only supports Wave files that are either uncompressed or compressed with the (lossy) μ -Law or a-Law algorithms.

The Wave format supports markers and multichannel data. **AmadeusPro** will try to do its best to map the multitrack structure of the document to the ones supported by the Wave format, but this may fail if your document contains too many different tracks.

Metadata (artwork, artist name, etc) are saved to Wave files by appending an ID3 tag at the end of the file. This is the method employed by iTunes and should not affect the readability of the resulting file by third-party software.

Due to the nature of the format, WAVE files are limited to 2GB in size. It is possible to encounter WAVE files that are between 2GB and 4GB in size. Such files *do not* comply to the official WAVE specifications and cannot be opened by every program supporting the WAVE format, this is why **AmadeusPro** will refuse to create them. It is however able to open such files and to save them in other file formats.

3.3 The AIFF format

This format is very popular mainly on the MacOS platform. It is similarly structured to the Wave format and usually contains uncompressed data, but also supports a variety of compressed formats.

Like the WAVE format, the AIFF format supports markers and multichannel data. **AmadeusPro** will try to do its best to map the multitrack structure of the document to the ones supported by the AIFF format, but this may fail if your document contains too many different tracks. Metadata (artwork, artist name, etc) are also saved to AIFF files by appending an ID3 tag at the end of the file.

Due to the nature of the format, AIFF files are limited to 2GB in size. The same remark as for WAVE files applies to AIFF files of sizes between 2GB and 4GB.

3.4 The QuickTime Movie format

Supports a variety of compression formats. When a multitrack document is saved as a QuickTime Movie file, **AmadeusPro** will downmix it to a mono or stereo file, whichever is more suitable. Metadata is lost when saving to the QuickTime Movie format.

Warning: At the moment, **AmadeusPro** does *not* keep video tracks in QuickTime movies. If you wish to edit the sound track of a video clip, you should save the new sound track into a separate file and then use a tool like iMovie to merge it back into the clip.

3.5 The NEXT/Sun format

A popular sound format on the Linux platform. **AmadeusPro** only supports NEXT/Sun files containing uncompressed sound. It will downmix multitrack documents it to a mono or stereo file, whichever is more suitable. Metadata is lost when saving to the NEXT/Sun format.

3.6 The AAC ADTS, MPEG-4, and Bookmarkable AAC formats

These are different container formats that are primarily used to save data compressed with the lossy AAC compression algorithm. **AmadeusPro** will downmix multitrack documents it to a mono or stereo file, whichever is more suitable.

Metadata is lost when saving to the AAC ADTS format, but it is preserved when saving to the three other formats.

Note that the AAC ADTS format is *not* the same as the Mp4 or Apple's M4a format. These are three different container formats and QuickTime for example will not be able to read AAC ADTS files. On the other hand, the M4a and the Bookmarkable AAC format differ only in the file extension, but are treated differently by iTunes and the iPod.

In most cases, you will probably wish to use the **MPEG Audio (.m4a)** format as this is the format used by Apple's iTunes Store.

3.7 The Apple CAF File format

This is a format developed by Apple and designed to replace AIFF in the long term. It removes the 2GB file size limitation of AIFF and supports a wider variety of compression algorithms.

While the CAF file format does in principle support sound data with many channels, the current version of **AmadeusPro** only supports the reading and writing of mono or stereo files in this format. **AmadeusPro** will therefore downmix multitrack documents to a mono or stereo file, whichever is more suitable.

Metadata is lost when saving to the Apple CAF format (this may change in the future, as the format in principle supports the inclusion of metadata).

3.8 The FLAC file format

FLAC (Free Lossless Audio Codec) is a compressed lossless sound format popular on the Linux platform. It is possible to choose between different compression levels. Since the format is lossless, there is no downside in principle to choosing the highest possible compression, but it may make file saving operations significantly slower for a minor gain in file size.

When a multitrack document is saved as a FLAC file, **AmadeusPro** will downmix it to a mono or stereo file, whichever is more suitable. Most text-based metadata is preserved in the FLAC file format, but artwork is lost.

3.9 The Ogg Vorbis format

This is a popular alternative to the Mp3 format, especially on the Linux platform. At intermediate bitrates (128–256kbps), most listening tests favour Ogg Vorbis over Mp3 in terms of quality. From an end-user's perspective both formats are very similar in nature and most of what is said for the Mp3 format below also applies to the Ogg Vorbis format. In particular, the settings are almost identical.

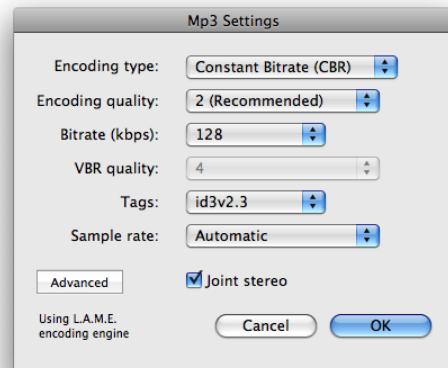
When a multitrack document is saved as an Ogg Vorbis file, **AmadeusPro** will downmix it to a mono or stereo file, whichever is more suitable.

Vorbis is the first of a planned family of Ogg multimedia coding formats being developed as part of [Xiphophorus's Ogg multimedia project](#).

3.10 The Mp3 file format

This is an extremely popular lossy sound compression algorithm. **AmadeusPro** reads Mp3 (and also Mp2) files through the tool mpg123 (see Section 15 below).

Saving Mp3 files is achieved through the tool `lame`. (See also Section 15 below.) It supports a variety of settings (the meaning of the most important one, the bitrate, was explained above). In order to achieve a given file size, it is recommended to use the **ABR** (Average Bit Rate – simple parts of the sound will be compressed more than complex parts, achieving a better overall quality) setting, rather than **CBR** (Constant Bit Rate – all parts of the sound have the same compression rate) in order to achieve maximum quality, but note that some Mp3 players may not support ABR files. The **VBR** (Variable Bit Rate) setting is more flexible and allows you to specify a target quality rather than a target file size. This usually results in better sounding files, but you have less control over the resulting file size ('complex' sounds will produce larger files than 'simple' sounds for a given quality). Note also that as with ABR files, not every Mp3 player supports VBR files. The screenshot next to this paragraph shows a popular setting that represents a good trade-off between file size and sound quality. Note that the use of id3v2.4 tags is discouraged since their support is broken in iTunes (this is true as of iTunes 7).



The **Sample rate** setting allows you to resample the sound before encoding. In most cases, the **Automatic** setting will yield best results. For example, when encoding a 44.1kHz AIFF file at 32kbps, `lame` will resample it to 16kHz. This will in general yield better result since less bandwidth is wasted on high-frequencies and so the lower frequencies can be described more accurately. By changing this setting to **Original**, this behaviour can be suppressed.

When a multitrack document is saved as an Mp3 file, **AmadeusPro** will downmix it to a mono or stereo file, whichever is more suitable. Metadata (including artwork) is preserved in Mp3 files, provided that you choose to create an id3v2 tag.

The remaining settings are mainly for expert usage. If you do not know what they mean, then you will probably not need them.

3.11 The SoundDesigner format

This used to be a popular format on the Macintosh platform and it is still the format of choice for the software Digital Performer. It can only contain uncompressed sound data, but has the advantage over AIFF and Wave of not having a 2GB file size limitation.

3.12 About raw data

AmadeusPro is able to import and export sound as raw sound data. Unless you are trying to recover data from corrupt files or know precisely what you are doing, you are unlikely to need this functionality.

In particular, please do *not* use the **Import Raw Data** function to open normal sound files (AIFF, Wave, Mp3, etc) unless you really do know what you are doing. Results are likely to range somewhere between mild annoyances and total garbage.

If you are using the **Import Raw Data** function in order to recover data after a crash of **AmadeusPro**, you should make sure that the **Header size** field is set to **0** and that the **Samples Format** field shows the correct sample size, as well as the type of computer (Intel or PowerPC) that you are currently using.

3.13 Exporting markers

The **Export Markers...** function of the **File** menu allows you to export the markers of the currently open document as a [.cue file](#). This is a text-based file format that is commonly used by CD burning programs. It has the advantage of being editable by any text editor, so that it is also possible to create your own .cue files ‘by hand’. The **Import Markers....** function allows you to open .cue files and to add the corresponding markers to the current document.

3.14 iTunes support

AmadeusPro plays nicely with iTunes. The **File** menu has an **Export to iTunes library...** item that allows you to add the open document to your iTunes music library. It is recommended to set the metadata (song name, artist name, etc) prior to selecting this function, so that the song shows up properly in iTunes.

Next to this, there is a **Export as iPhone Ringtone...** item that does pretty much the same thing, but instead of adding the document to your music library, it adds it to your ringtone library. Both of these functions were only tested with iTunes 8. They might not work correctly with earlier versions.

4 Sound recording and playback

4.1 Sound recording

The first step for any recording is of course to set your hardware up in such a way that the sound gets into your computer. Most Macintosh models have a line-in audio input built in, so you can plug the output of your Hi-Fi system directly into it (you will just need a standard [mini-jack](#) cable). If this is not the case for your computer (check with your local Apple store if you are unsure about which one is the line-in plug) or if you wish to obtain better control and / or better sound quality, you can use one of many third-party audio devices. Any audio device compatible with MacOS X should be recognized correctly by **AmadeusPro**. Popular brands include [Griffin’s iMic](#), [Motu](#), [Roland / Edirol](#), [Digidesign](#), etc. I am not affiliated with any of these manufacturers and I can therefore not provide any advice on comparing their products or using them.

Once your hardware is set up, the easiest way of recording a sound with **AmadeusPro** is to click on the red **Record** button depicted here. By default, this is exactly the same as choosing **Record...** in the **Sound** menu. This will bring up a standard recording dialog and start the recording immediately. It is possible to change this behaviour, see Section 10. Once your recording is complete, just hit **Return**

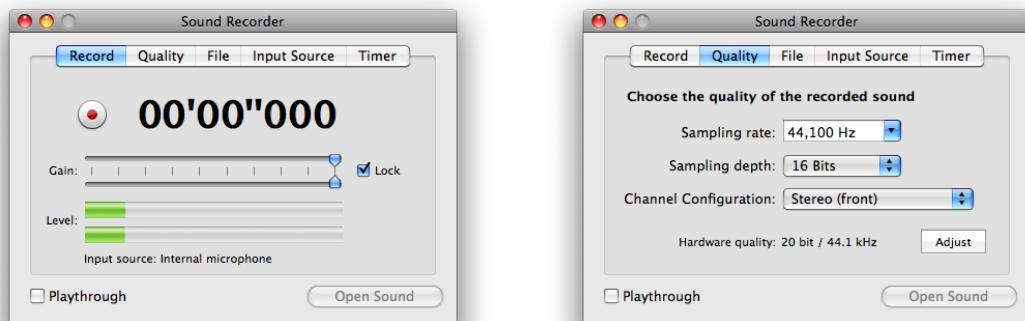


or click the **Stop** button and the recorded sound will be inserted at the insertion point in the current track. You can also have **AmadeusPro** create a new track containing the recorded sound by choosing the **Record to New Track...** function instead. The function of the **Record** button can be customized by clicking on it and leaving the mouse button pressed for half a second. This will show a pop-up menu that allows to choose between the **Record...**, **Record to New Track...** and **Record to New File...** function. This choice will be remembered next time the **Record** button is clicked.

The standard recording dialog only provides control over the volume of the recording. The volumes of both channels (for a stereo recording) can be adjusted individually. Setting the **Lock** checkbox forces **AmadeusPro** to use the same volume for both channels. Note however that **AmadeusPro** is able to provide volume control only if your hardware allows it. Some sound recording devices do not provide any control over the recording volume and in this case, the corresponding sliders will be grayed out. Please do not report this behaviour, as it does not constitute a bug. **AmadeusPro** does intentionally not provide software volume control at the time of recording since this can only diminish the resulting sound quality. If you are not able to adjust the volume correctly in the recording window, you should make sure that your hardware is set up properly.

One very common mistake is to plug a microphone directly into a [line-in](#) audio input. Without a [preamplifier](#), this will result in extremely low sound volume and poor sound quality. The iMic for example provides a built-in [preamplifier](#) that can be controlled with a little black switch. A similarly common mistake is to plug the line-out output of a home Hi-Fi system into a microphone input. This will result in a very saturated sound. Both of these problems can only be solved by setting up the hardware properly and **not** by any software solution. If you are unsure about how to connect your sound source to your computer, please seek advice from your nearest Apple store or from the manufacturer of your audio input device.

Complete control over sound recording can be achieved by choosing **Record to New File...** in the **Sound** menu. This brings up the following dialog:



The **Record** tab provides the same functionality as described above.

The **Quality** tab allows you to specify the sampling rate and sampling depth of the resulting sound. Clicking on the **Adjust** button allows you to attempt to adjust the hardware quality to match the selected quality as closely as possible. If it is not possible to match the selected quality exactly, **AmadeusPro** will automatically convert the recorded sound to the selected quality. This tab also allows you to choose which channels to record from. From the point of view of **AmadeusPro**'s recording function, every sound input device has 5 channels: front left, front right, back left, back right and center. Of course, your particular sound input device may have more or (more likely) less input channels than that. It is therefore possible to assign the

physical channels of your input device to the 5 channels of **AmadeusPro**'s recording engine by clicking on **Advanced settings** in the **Input Source** tab.

The **File** tab allows you to save a copy of the recorded sound while it is being recorded. This allows you, for example, to record a radio show directly as an Mp3 file, therefore not having to spend a long time encoding the recording once it is completed. **Warning:** If you select a configuration with more than two channels in the **Quality** tab, only the first two channels will be saved to the file.

It is not necessary to use this function as a safeguard against the unlikely event of **AmadeusPro** crashing during a recording: the recorded sound is always saved into a temporary file. This file will be placed in the trash bin by **AmadeusPro** next time you launch the program and it can be recovered with the **Import raw data** function.

The **Timer** tab allows you to schedule a recording for a predetermined time of the day. If you choose the recording to start after a certain amount of time, you need to press the **Record** button in the **Record** tab to launch the timer.

The recording window obeys to a number of handy key shortcuts that are summarised in the following table:

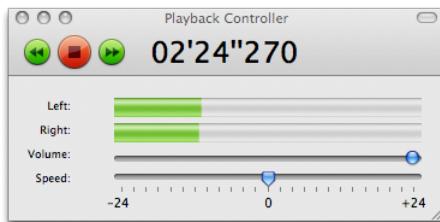
key	action
space	Starts / stops the recording
'M'	Creates a marker at the current location of the recording
'm'	Same as 'M', but a dialog comes up, allowing to edit the marker
tab	Same as 'M'.

4.2 Sound playback

Once a sound file has been opened in **AmadeusPro**, the most common task is to play it back. In a nutshell, playback is controlled by the spacebar. Hit the spacebar to start it and hit it again to stop it. When no sound is playing, hitting the tab key is equivalent to first hitting the **rewind** button and then hitting the space bar. When a sound is playing, hitting the tabulation key is equivalent to first hitting the space bar and then hitting the **rewind** button.

The current playback position is indicated by the playback head, which is displayed in the timescale of the main document. The playback head is depicted on the right. The location of the playback head can be changed by simply clicking at the corresponding location in the timescale. Whenever you click in the main sound window and the sound is not being played back, the position of the playback head is reset to the position selected in the **General** pane of the **AmadeusPro** preferences dialog. This position would be either the start of the current selection or the start of the document. The precise behaviour of the space bar is the following. If no sound is currently being played back, playback starts at the location of the playback head. If a sound is being played back, playback stops. If **Space bar rewinds sound** is checked in the **General** pane of the **AmadeusPro** preferences dialog, then the position of the playback head is reset, otherwise it stays where it was when the space bar was hit.

More sophisticated control can be achieved through the playback controller:



The **Play** icon has exactly the same functionality as hitting the spacebar. The **Fast forward** and **Fast rewind** icons allow you to scrub through a sound while it is playing back. The **Speed** slider allows you to adjust the playback speed. The scale is measured in semitones (12 semitones make up one [octave](#)) so that maximal playback speed is double the original speed and minimal playback speed is half. The **Volume** slider allows you to adjust the overall volume of the sound being played back.

4.3 Adjusting the playback device

The sound output device used for playback can be selected in the **Sound** preferences panel accessible by choosing **Preferences...** in the **Amadeus Pro** menu.

Like the situation for recording, from the point of view of **AmadeusPro**'s playback function, every sound input device has 5 'virtual' channels: front left, front right, back left, back right and center. It is possible to assign the physical channels of your input device to the 5 virtual channels by clicking on **Advanced settings** in that preferences dialog. On a normal two-channel output device (built-in speakers for example) the virtual center channel is assigned to the first available physical channel, which is usually the left speaker. Therefore, a mono track set up to play back on the center channel has a good chance of being played back through your left speaker. If you wish it to be played on both speakers simultaneously, you should select **front or back** in the **Playback** menu of the track.

Note that by default, the output device used for playback is the one selected in the **Sound** system preferences dialog. However, if you set a playback device using **AmadeusPro**'s preferences, this takes precedence over any settings that you may have in the system preferences. The setting will also be remembered from one session to the next.

5 Editing

5.1 Navigation

The first thing anyone would like to do when editing a file is of course to find one's way around the file. The two most obvious ways of navigating through a sound are the **Horizontal zoom** slider that allows you to change the timescale and the horizontal scrollbar located just below the timescale.

However, there are many other ways to finding your way around a document. These are:

- The **Markers** palette already mentioned in the previous section.
- The '+' and '-' key shortcuts that allow to zoom in and out of the document.

- The Zoom In / Zoom Out items in the toolbar (if they are not there and you happen to need them, just choose **Customize Toolbar** in the **View** menu). There are corresponding menu items in the **View** menu.
- The **Zoom to Selection** item in the **View** menu changes the display in such a way that the selected portion of the current track just about fills up the display. If the selection is empty, the waveform is expanded to the maximum (1sample = 1pixel) and it is moved in such a way that the insertion point is located at the center of the display. The **Zoom to Document** item in the same menu has a similar effect, but changes the zoom level so that the entire document neatly fits into the window.
- The scroll wheel of your mouse also allows you to zoom in and out of a document. If your mouse has a horizontal scroll wheel as well (like Apple's Mighty Mouse or the 'two finger scrolling' feature on the trackpads of recent laptops) you can use it to move the document horizontally. There is a hidden preference that allows you to exchange the behaviour of the horizontal and the vertical scroll wheel, see Section 10.
- The **Move to Selection** and **Move to Playback** menu commands move the display (without changing the timescale) in such a way that the center of the selection (or the playback head) is at the center of the display.
- The keys '1' to '4' allow you to set the zoom level to one of four predefined values. You can set the value corresponding to a key to the current zoom level by typing 'Option-1', 'Option-2', etc.

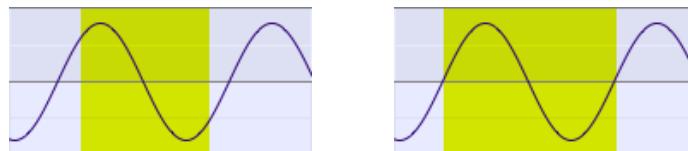
5.2 Basic editing

Most of the basic editing functions (Copy / Paste / Cut / Delete) work in exactly the same way as in any text editor or image editor. The **Delete in All Tracks** function works like **Delete** but extends its action to all the tracks of the document instead of just the currently selected one.

The **Paste Over** function allows you to literally paste the content of the clipboard over the current sound, effectively mixing it with the document instead of just inserting it at the insertion point.

Another useful feature of **AmadeusPro** is that tracks can be shuffled around using drag and drop, by clicking into the shaded grey track header. In particular, they can be moved from one document to another. If the sampling rate and / or sampling depth of the dragged track does not match that of the receiving document, it will be converted accordingly.

The **Extend to Crossing** function in the **Selection** menu allows you to extend the current selection to the nearest points where the waveform crosses the 0-line. This is probably best illustrated by the following two screenshots, where the left hand side shows the selection of a sound before applying this function and the right hand side shows the same sound after applying the **Extend to Crossing** function.



This is very useful in order to avoid little crackling noises at the points where sounds are cut and pasted.

5.2.1 Inserting silence

To insert a silent portion of a given length, use the **Generate Silence...** function from the **Effects** menu. To silence a portion of a sound, first select the portion you wish to silence and then choose **Amplify** from the **Effects** menu, with a setting of 0%. (Make sure that % is selected and not dB!) This has the advantage that you can optionally choose a short fading, which will make for a smooth transition to the silent portion instead of an abrupt transition which may sound like a crack.

5.2.2 Changing the characteristics of a sound

The characteristics (sampling rate and bit depth) of a sound can be changed by choosing **Characteristics...** in the **Sound** menu. If the “don’t resample” checkbox in that dialog is checked, the current sound will simply be played back at a different speed, without changing the actual sound samples.

For example, if you change the sampling rate from 44.1kHz to 22.05kHz and have “don’t resample” checked, the sound will be played back at half speed (so its pitch will also be lowered by one octave). If the “don’t resample” option is left unchecked, then half of the sound samples will be thrown away, so that playing back the sound at half speed results in something sounding roughly identical (except for a loss in the high frequencies) to the original.

5.3 Applying effects to a sound

AmadeusPro ships with a number of sound effects covering most needs, but its capabilities can be greatly extended by the addition of third-party effects. **AmadeusPro** supports effects in both Steinberg’s VST and Apple’s Audio Unit format. However, Audio Units are preferred over VST effects. The support for VST effects is frozen at the current state whereas Audio Unit support is expected to improve in future versions of **AmadeusPro**.

Several Audio Units are shipped by Apple with MacOS X and should show up in the **Audio Units** submenu without having to take any action. Installing a third-party effect is as simple as dragging the corresponding file into the appropriate subfolder (**Components** for Audio Units and **VST** for VST effects) of your ‘Library/Audio/Plug-Ins’ folder. A collection of third-party Audio Units can be downloaded [here](#).

In order to apply a sound effect to a portion of a sound, select this portion first and then select the desired sound effect. In order to apply an effect to the whole track, you can either leave its selection empty or select the whole track with the **Select All** function from the **Edit** menu.

The remainder of this subsection is devoted to the description of the content of the **Effects** menu. The description of the **Interpolate** and **Denoising** items is however postponed to Section 8.

5.3.1 Echo

Applies a simple echo to the sound (this effect is also called ‘Delay’ by some audio programs). The **gain** setting describes the intensity of the echo, it should always be a negative amount in dB or a percentage less than 100%. The **delay** setting gives the delay of the echo, which is the amount of time after which the sound is repeated. The **further** setting, when the corresponding checkbox is ticked, allows to extend the echo

further than the selection. To appreciate its effect, try selecting a drumbeat and apply an echo with a delay of 0.2 seconds, a gain of 60% and the **further** setting set to 2 seconds.

5.3.2 Amplify

Allows to amplify (or decrease) the volume of the selection by a given amount. See the box on page ?? for a discussion of the meaning of the dB and % units. The **Fading** function allows to make a smooth transition to the selected volume at the start and end of the selection.

5.3.3 Equalizers

The **Equalizers** submenu contains three standard multi-band **equalizers** (they work in exactly the same way as the graphic equalizers on your hifi equipment), as well as a collection of specialized equalizers aimed at the restoration of old vinyl recordings. These specialized equalizers are described in this section.

The **RIAA Reproduction** filter allows you to compensate for the standard RIAA filter usually applied on vinyl recordings. See for example <http://www.bonavolta.ch/hobby/en/audio/riaa.htm> for more information. In general, vinyl recordings are engraved with reduced bass levels and increased treble levels. In 1954, the **RIAA** (Recording Industry Association of America) introduced a standard filter that was used on nearly all phonograph recordings since then. The **RIAA Reproduction** filter allows you to apply the RIAA curve to a recording. This is usually done by the turntable preamplifier. It may however be useful if you connect a turntable directly to your computer or use a normal preamplifier.

The **Inverse RIAA Reproduction** filter applies the inverse of the RIAA curve to a recording. This is useful for undoing the effect of a turntable preamplifier on acoustic LP recordings that were not engraved with the RIAA curve.

The **RIAA/IEC Reproduction** and **Inverse RIAA/IEC Reproduction** filters work in a similar way, except that they use a modification of the original RIAA curve that was introduced around 1976 but is not as widespread.

The **LP / 78rpm Equalizer** is a parametric equalizer that is designed to allow the reproduction of old vinyl recordings. It contains a number of presets for leading manufacturers. More information can be found for example at

http://www.hal-pc.org/~clement/Restoring%20Old%20Recordings.htm#Appendix_B

If the recording is made through a standard turntable preamplifier, **RIAA de-emphasis** should be enabled in order to compensate for the RIAA curve.

5.3.4 Fadings

The **Fadings** menu contains various functions that allow to change the volume of a sound. The fading curves used by the **Fade in** and **Fade out** functions can be selected in the palette that opens when selecting the **Settings...** menu item.

The **Symmetric fade** function is equivalent to first applying a **Fade in** to a portion of the desired length at the beginning of the selection, followed by applying a **Fade out** to a portion of the same length at the end of the selection.

The **Ramp** function allows to change the volume of the sound from one level at the start of the selection to another level at the end of the selection.

The **Crossfade to the Left** and **Crossfade to the Right** functions allow you to easily create a transition between two pieces of sound. This is how it works. Let's say you want to produce a file containing the songs 'Song A' and 'Song B'. Copy them one behind the other into your file and place a marker at the junction of both songs. Then select, say 1 second of 'Song B' starting from the marker and select **Crossfade to the Left**. This will first create a fade in of 1 second at the beginning of 'Song B' and a fade out of 1 second at the end of 'Song A'. It will then merge these two parts into one, creating a nice transition. The **Crossfade to the Right** produces the same result if you select 1 second at the end of 'Song A'.

The **Loop Crossfade** function is similar to what was just described, except that it creates a transition between the end and the start of the selection. This allows the resulting sound to be looped seamlessly.

5.3.5 Normalizing a sound

A commonly performed task is to bring the level of a recording to some standard volume level. This can be performed through the **Normalize...** command. A problem with this is the notion of 'level', as there are many different ways of measuring a sound's level. The **Normalize...** function supports two notions of sound level. The **maximal sample value** simply looks for the sample that has the largest absolute value and then amplifies the whole sound in such a way that this sample is brought to the reference level. This has the advantage that if the level is chosen to be smaller than 0dB, it is guaranteed that no clipping will occur. The disadvantage is that a short outburst of unusual loudness or a crackle in the sound will dominate the picture, so that the achieved effect is not always what was intended.

The **maximal RMS power** setting tries to compensate for this by taking average levels over relatively short time intervals (100ms) and choosing the largest of these average levels as the reference. This has the advantage of being usually closer to the desired effect. However, it does not guarantee that no clipping will occur. Normalizing the **RMS power** to around -10dB or -15dB is quite safe in most situations.

Note that the **Normalize...** function adjusts the global volume of the sound, but does not compensate for variations of volume within a recording. If you wish to obtain several songs sounding very homogeneous throughout (as required for example for background music in a store), you may want to apply dynamical compression before normalizing. One sequence of steps that works well for most situations is to first apply Apple's **AUDynamicsProcessor** Audio Unit with the 'hard' factory preset and then to normalize the RMS power to 25%.

5.3.6 Reverse

This effect reverses the time on the selection, thus playing it backwards.

5.3.7 Audio Units

As explained above, [Audio Units](#) are plug-ins developed by third parties that allow to extend [AmadeusPro](#)'s (and any other Audio Units host's) abilities. The **Audio Units Rack** allows to apply several Audio Units in one go. Simply drag the desired effects from the **Available Audio Units** section to the **Applied Audio Units** section of the rack window. In order to change the settings of an effect, either double-click on it or select it and click on the button with the little pen. You can remove an effect from the list of applied Audio Units by selecting it and hitting the 'backspace' key or clicking on the button with the minus sign.

The following list of effects ship with MacOS X 10.4. Unfortunately, Apple doesn't ship any documentation whatsoever for their settings so your bet is as good as anyone's. For some of the more esoteric settings, experimenting is probably the best option.

- **AUBandpass** Suppresses all frequencies outside of a specified band.
- **AUDelay** Same as the **Echo** effect. It includes a dry/wet mix setting that allows to choose the proportion between the original sound and the echo. At 0%, it makes no modification to the sound, at 100% it keeps only the echo and removes the original sound. The default value of 50% corresponds to the behaviour of [AmadeusPro](#)'s **Echo** effect.
- **AUDynamicsProcessor** This allows to expand or compress the dynamical range of a sound. The dynamical range is the ratio in volume between the quiet parts and the loud parts. Compressing the dynamical range results in the volume becoming more uniform throughout the sound. When normalized afterwards, the quieter parts will therefore sound louder. This effect is often used (and abused) by record labels in order to make a CD sound louder. The different presets shipping with this effect give good results.
- **AUFilter** This is basically a combination of an AUHipass and an AULowpass filter, as well as three parametric equalizers.
- **AUGraphicEQ** Standard 10 or 30-band graphical equalizer.
- **AUHighShelfFilter** Reduces drastically frequencies below a cut-off frequency. The gain setting allows to adjust the strength of this reduction.
- **AULowShelfFilter** The opposite of AUHighShelfFilter, it reduces drastically frequencies above a cut-off frequency. The gain setting allows again to adjust the strength of this reduction. Useful to remove a low rumble in a recording.
- **AUHipass** Pretty much the same as AULowShelfFilter.
- **AULowPass** Pretty much the same as AUHighShelfFilter.
- **AUMatrixReverb** Very sophisticated reverb effect. It takes stereo effects into account to create the impression of a large enclosed space and can therefore not be applied on mono tracks. You will probably want at least to lower the dry/wet mix to 50% from its default value of 100%.
- **AUMultibandCompressor** A compressor (like the AUDynamicsProcessor effect) that however first divides the sound into four separate frequency bands, determined by the 'crossover frequencies'. It then allows to have different settings for each of the four frequency bands.

- **AUNetSend** Allows to send the sound over a network. It doesn't perform any processing and is therefore only useful in **Preview** mode.
- **AUParametricEQ** Allows to boost or decrease the volume in a frequency band centred around the 'center frequency'. The 'Q' value determines the width of the frequency band.
- **AUPeakLimiter** Amplifies the sound by 'pre-gain', but makes sure that it does not get clipped. Works essentially like a compressor.
- **AUPitch** Allows to change the pitch of a sound without changing its speed. The change in pitch is given in 'cents'. There are 100 cents in a semitone and therefore 1200 cents in one octave. The remaining parameters allow to fine-tune the effect.
- **AUSampleDelay** Simply shifts the sound forward by the specified number of samples. Probably useless in **AmadeusPro**'s context of off-line processing.

5.3.8 Change Pitch and Speed

This effect allows you to change the pitch and the speed of a sound independently. It is very useful for example to constrain a recording to a certain length. The algorithm used is Apple's AUPitch Audio Unit, which provides professional quality even though it is included with every copy of MacOS X.

Increasing the quality results in a deterioration of processing speed and vice-versa. The adjustment of the pitch is done in semitones, *i.e.* +12 corresponds to increasing the pitch by one octave and -12 to decreasing it by one octave. If **lock** is selected, the speed increase/decrease exactly matches the pitch increase/decrease in such a way that the result is the same as what would be obtained by just playing the sound back at a different speed. In that case, the sound is simply resampled and does not go through the AUPitch effect.

5.3.9 Stereo Utilities

These are a couple of utilities useful for processing stereo sound. The **Balance** function allows to adjust the balance between the left and right channel and the **Swap channels** function allows to swap the left and the right channel. The **Invert phase** function changes the sign of the waveform. This is useful in order to reduce the amplitude of the 'center' in a stereo recording, which is how many 'Karaoke' effects work. In order to apply a 'Karaoke' effect to a stereo recording, you can:

- Select **Split track** in the **Sound** menu.
- Select the whole first track and apply the **Invert phase** effect.
- Select **Merge with Next Track** in the **Sound** menu.
- Select **Convert Track to Mono** in the **Sound** menu.

5.3.10 VST Effects

VST plug-ins work very much like Audio Units and an effort has been made to present coherent interfaces for both types of plug-ins. The **VST Rack** therefore works in exactly the same way as the **Audio Units Rack**.

One commonly encountered problem is that many VST plug-ins have not been ported to Intel yet and only run on PowerPC computers. If this is the case, an error message will be presented. An archive with some free VST plug-ins that however work only on PowerPC computers can be found [here](#).

5.4 Maintaining a list of commonly used actions

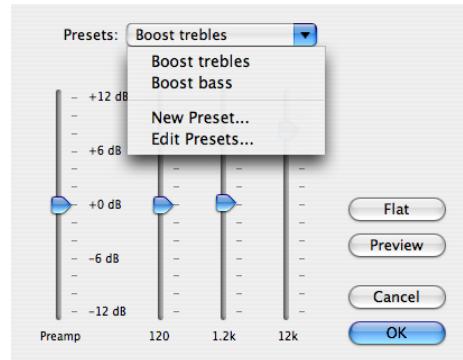
The first item of the **Effects** menu allows you to perform again the most recently performed action. The **Actions** tool accessible from the **Windows** menu maintains a list of the five most recently performed actions. It also allows you to save some favourite actions by simply dragging them from the **Last actions** list into the **Favorite actions** list. Favorite actions can be given any name by option-clicking on them. They can be deleted by moving them into the little trash bin in the lower right corner of the window. The **Clear last actions** button allows you to clear the list of last actions. It also clears the state of the first item of the **Effects** menu.

5.5 Working with presets

Most of the functions in **AmadeusPro** do not only remember their last settings, but allow you to maintain a list of presets which would typically be commonly used settings. For example, the window managing the 3-band equalizer is represented in the right margin.

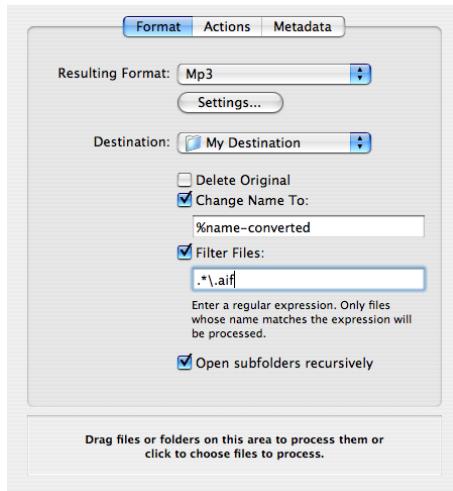
The **Presets** menu contains the list of presets with the name of the currently used preset being displayed inside the menu button. If these don't match any of the presets or if no preset exists, **Not a Preset** is displayed instead.

When you select **Save Preset...**, a dialog allows you to choose the name of the new preset, which will then be a copy of the current preset. The **Edit Presets...** function allows to manage the list of presets. In particular, it allows you to rename, to duplicate, and to delete them. Some (but by no means all) functions do have a default set of presets built into **AmadeusPro**. If this is the case, the presets menu also contains a **Revert to Defaults...** item which allows you to revert to the factory defaults in case you destroyed them by inadvertance. **Warning:** Your own presets will be lost in this case.



6 Batch processing

AmadeusPro includes a powerful [batch processing](#) engine. A new batch processor can be created by choosing **New Batch Processor** in the **File** menu. A batch processor allows you to not only convert files from one file format to another, but it also allows you to apply various effects (normalizing, fadings, etc). An example of a batch processor that converts AIFF files into Mp3 files could look like this:



The **Change Name To** field allows you to specify a name for the resulting file. The special sequence of characters ‘%name’ will be replaced by the name of the currently processed file. Other usable special sequences of characters are %title, %artist, %album and %year. These correspond to the contents of the corresponding metadata fields. However, note that these fields can be empty if the sound being processes contains no metadata.

The **Filter Files** setting allows you to enter a regular expression against which filenames will be checked. Only files with matching filenames will be processed. Explaining in detail what regular expressions are goes beyond the scope of this manual. Information about the type of regular expressions used by **AmadeusPro** can be found at the following link:

<http://icu.sourceforge.net/userguide/regexp.html>

Examples of valid regular expressions are

***.\.mp[2-4]** Matches any filename ending in ‘.mp2’, ‘.mp3’ or ‘.mp4’.

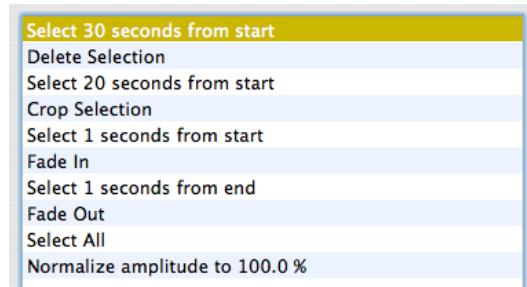
Track.* Matches any filename starting with ‘Track’.

A file or a collection of files is processed by the batch processor by dragging the file(s) onto the indicated area. If a folder is dragged to this area, only the files at the first level within the folder are processed by default, unless you check the **Open subfolders recursively** option, in which case the whole folder hierarchy is processed.

The **Actions** tab allows you to choose an arbitrary number of sound effects to be applied to the sound being processed. These effects are applied to the sound in the order in which they appear in the list. They are the same as the ones appearing in the **Effects** menu, except for a few items that allow to manipulate the current selection of the sound, as well as the sound quality. Suppose for example that you wish to create a 20 seconds summary in the Mp3 format for a collection of audio tracks. In order to achieve this, you would create a batch processor with the following settings:

1. In the **Format** section, select ‘Mp3’ as your resulting format and select the appropriate settings (see Section 3).
2. Check the **Change Name To:** checkbox and enter ‘%name-summary’ as your filename.

3. In the **Actions** section, add one **Set Selection...** action and choose 30 seconds from the start in the dialog that will appear. Then, add a **Delete Selection** action, followed by another **Set Selection...** action with this time 20 seconds from the start and a **Crop Selection** action. This leaves us with a 20 seconds piece of sound located 30 seconds into the original sound. We then add 1 second fade in and outs, followed by a normalisation. This should leave you with the actions section looking like on the following screenshot:



It now suffices to drag your collection of files onto the area at the bottom of the batch processing window and sit back to watch **AmadeusPro** do all the hard work for you.

The **Metadata** tab allows you to change the metadata of the sound being processed. This allows you, for example, to convert all the tracks from a CD to Mp3 and simultaneously set the artist and album names to the correct values. The special sequences of characters mentioned above can be used in any of the metadata text fields. So you can for example enter ‘%title - %track’ in the **Title** field in order to change a name like ‘Wish you were here’ to something like ‘Wish you were here - 3’.

7 Managing tracks

As we already saw in Section 2.4, an **AmadeusPro** document can consist of several tracks and each of them can be either mono or stereo. We have already seen in Section 5 how to edit the content of individual tracks. This section shows how to edit a document at the track level.

7.1 Moving and reordering tracks

Tracks do not have to start at time 0 but can be moved to an arbitrary starting time. To do so, just click on one of the small handles located at either end of the track (see the screenshot to the right) and drag it to the desired location. However, there always has to be *at least one* track starting at 0, so that the handles are greyed out if moving them would break this rule.

The order of tracks can also be changed and tracks can even be moved from one document to another one. To do so, click into the shaded gray area located at the top of the track and drag the mouse to the desired location. This location can also be within a different document.

7.2 Creating and deleting tracks

The items of the toolbar showed to the right allow to manage tracks. The first one creates a new track with the same characteristics as the current track. This can also be



achieved with either the **Add New Mono Track** item or the **Add New Stereo Track** item in the **Sound** menu. The second toolbar item deletes the current track, which can also be done via the **Delete Track** menu item. The third toolbar item allows to split a stereo track into two mono tracks, one for each channel. This corresponds to the **Split Track** menu item. It is very useful in situations where the two channels of a stereo track really correspond to two completely different parts of a recording (like the vocals and the background music or the two participants in a Skype conversation) and should be edited separately. The **Merge With Next Track** menu item is the opposite of the **Split Track** function and allows to combine two mono tracks into one stereo track.

Most track-related functions described above can also be accessed by right-clicking (control-clicking on a one-button mouse) into the shaded gray area at the top of a given track.

8 Sound Restoration

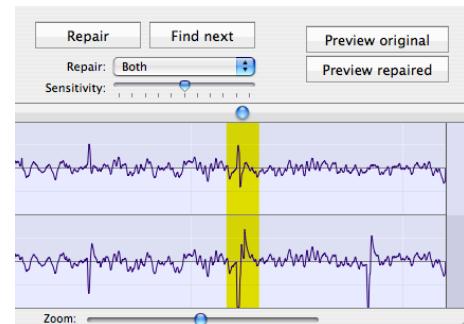
One of the strengths of **AmadeusPro** is that it provides you at low cost with two tools to restore damaged recordings that are of a quality that rivals that of professional products. These two tools have two very different purposes: either the elimination of very short and precise cracks and pops or the elimination of a continuous background noise.

8.1 Elimination of pops and cracks

A common problem of old vinyl recordings are the pops and cracks that are caused by scratches in the disk. Provided that they are isolated and that there are not too many of them, such cracks can be eliminated in a very efficient way by **AmadeusPro**.

In order to eliminate a single isolated crack, proceed as follows. First, find the location of the crack in your sound and zoom into it. A crack can usually be spotted visually as a spike in the waveform. Then, select the entire crack and choose **Interpolate** in the **Effects** menu. Alternatively, you can also just hit the ‘r’ key. Make sure that you select a piece of sound that is *as short as possible*. Ideally, you want to select only one isolated crack and nothing else. If a too long piece of sound is selected, **AmadeusPro** will complain and refuse to apply this function.

If a sound contains many cracks, and in order to have more control over the repairing, you can use the **Repair Centre** tool accessible from the **Windows** menu. This shows a zoomed display of the waveform of the current track. One convenient feature of this display is that it is always scaled vertically in such a way that the whole height of the display is being used. This allows you to see the waveform accurately even if it has a low amplitude. The small blue knob at the top of the display allows you to scroll through the waveform, the position of the knob determines the speed of the scrolling. If a location in the waveform looking like a crack is found, you can determine whether it really is a crack in the following way. First click on **Preview original** which plays a short section of sound (about one second) containing the current selection. If this section contains a crack, click on **Preview repaired** which plays exactly the same section, but with the selected portion repaired. If this provides a satisfactory result, click on **Repair** in order to apply the repair to the sound. If it doesn’t produce any audible improvement, it may be possible that the real crack is located slightly after or before the current location.



The **Find next** function looks for the first crack following the current location. This function may return false positives (locations that look like cracks to **AmadeusPro** but that are in fact not), so you should always manually verify any crack found using this function.

The repair centre window obeys to a number of handy key shortcuts that are summarised in the following table:

key	action
r	Repairs the current selection
shift-r	Repairs the current selection and finds the next crack
t	Repairs the top channel of the current selection
shift-t	Repairs the top channel of the current selection and finds the next crack
b	Repairs the bottom channel of the current selection
shift-b	Repairs the bottom channel of the current selection and finds the next crack
o	Preview original sound
p	Preview repaired sound
n	Finds the next crack
z	Undoes the last action
shift-z	Redoes the last action

The basic sound repair functions are also available with simple key shortcuts from the main sound window, even if the

8.2 Elimination of a continuous background noise

Another common problem is the presence of a continuous background noise on a recording. One example is the background hiss on a recording coming from a magnetic tape. Another example is the 50Hz or 60Hz hum coming from the electric grid. (This is commonly picked up by microphones that are somehow connected to a device that runs on a power supply and not on battery.)

Warning: However good a software package is, it cannot resurrect what is lost. You should therefore always try first to get the best possible quality by choosing appropriate recording equipment and only use software-based restoration functions as a matter of last resort. If a recording is important to you and cannot be repeated, try to make tests in similar conditions first.

The **Denoising** function in the **Effects** menu provides a very efficient algorithm for suppressing background noise. The easiest way to use it is the following:

- Select a piece of your sound containing *only noise*.
- Select the menu item **Sample noise**. The plugin will scan the selected noise and construct a noise profile.
- Select the whole piece of sound.
- Select the menu item **Suppress noise**. This will remove the background noise from the selected part of the sound, according to the noise profile constructed previously.

Sometimes, there is no piece of sound containing only noise available. In this case, you may want use the **Suppress White Noise** function. The easiest way to proceed is to use the **Preview** function to play the selected sound and then to adjust the **Noise RMS power** with the slider provided until the result is

satisfactory. Note that choosing too large a value of the **Noise RMS power** will result in a significant loss of quality in the clean signal, whereas choosing a too small value will leave some background hiss remaining.

It may happen sometimes that the background noise one wishes to remove is very well localised in frequency (like the case of the hum discussed above). In this case, it is advisable to use the **Suppress Frequency Band...** function to remove every frequency localised between the **Lower Frequency** and the **Upper Frequency**.

Finally, the **Suppress DC Offset** function allows you to remove a low-frequency drift in the microphone (this happens for example on the integrated microphone of some of the older PowerBook models).

The **Denoising** function is customizable with the **Settings...** menu item. The **Noise Type** allows you to choose the type of noise you want to remove. Choosing a peaked noise is suited for background noise which is very well localised in frequency space. A typical example is the removal of the 50Hz or 60Hz humming produced by the power supply. The **smooth noise** function is adapted to the removal of noise that is spread over the whole frequency range, like for example a tape hiss. If the **Adaptive Filtering** option is disabled, the program will compute one global filter and apply this filter to the whole sound. This sometimes yields better results when the signal-to-noise ratio is very low. The **Sensitivity Enhancement** option tells the program the amount of background noise to remove. Usually, values between 30% and 70% yield the best results. For lower values, some background noise may remain after the application of the filter. For higher values, the signal may be substantially altered.

The **Algorithm** pop-up menu allows you to choose which [algorithm](#) should be used for the denoising. Currently, two algorithms are available: **Short FFT** and **Long FFT**. Generally, the **Long FFT** algorithm produces better results, but the **Short FFT** algorithm is slightly faster.

9 Key shortcuts

First of all, note that it is possible to change the key shortcuts for menu items in the **Keyboard Shortcuts** section of the **Keyboard & Mouse** system preferences pane. This is an extremely useful feature that is often overlooked by people and works in every application on MacOS X.

Here is a list of the key shortcuts that can be used in the main sound window of **AmadeusPro**:

key	action
space bar	Starts sound playback
command-space bar	Starts playback of the current track only
tab key	Rewinds and then starts playback
option 1-4	Stores the current zoom level into the corresponding register
1-4	Sets the current zoom level the value stored in the corresponding register
e	Previews the effect of deleting the current selection
a	Extends the selection by one pixel to the left
s	Shrinks the selection by one pixel at the left
d	Shrinks the selection by one pixel at the right
f	Extends the selection by one pixel to the right
g	Toggles the display of the grid behind the waveform
h	Toggles the display of the selection in the timescale
n	Cycles the currently active track between all the tracks of the document
m	Creates a marker at the insertion point
r	Repairs the selection (select only a very short piece)
t	Repairs the top channel of the selection (select only a very short piece)
b	Repairs the bottom channel of the selection (select only a very short piece)
p	Creates a marker at the playback head
down arrow	Plays back half an octave lower
up arrow	Plays back half an octave higher
left arrow	Places the insertion point at the left end of the selection
right arrow	Places the insertion point at the right end of the selection
[Places the insertion point at the left end of the selection. If nothing is selected, moves the insertion point to the left by one pixel.
]	Places the insertion point at the right end of the selection. If nothing is selected, moves the insertion point to the right by one pixel.
option-left arrow	Moves insertion point to the previous marker
option-right arrow	Moves insertion point to the next marker
command-left arrow	Extends the selection to the previous marker
command-right arrow	Extends the selection to the next marker
home	Jumps to the beginning of the sound
end	Jumps to the end of the sound
page up	Moves to the left by one screen
page down	Moves to the right by one screen
+ or =	Zooms deeper into the sound
-	Zooms out of the sound
option-‘+ or =’	Zooms vertically into the sound
option-‘-’	Zooms vertically out of the sound
command-click	Places a marker at the clicked point
shift-click	Extends / shrinks the selection to the clicked point
double-click	Starts playing the current track from the clicked point

10 Hidden preferences

There are a few preference settings that cannot be accessed from the user interface. They can be set from the Terminal.app application by typing a command like

```
defaults write com.HairerSoft.AmadeusPro HSMYSetting value
```

where `HSMYSetting` is the name of the setting in question and `value` is the value to which you wish to set it. The list of hidden settings is given in the following table, where the **Default** column shows the value corresponding to **AmadeusPro**'s default behaviour.

Name	Meaning	Default
<code>HSInvertScrollingBehaviour</code>	A value of 1 tells AmadeusPro to use the scroll wheel to control navigation rather than the zoom level.	0
<code>HSNumberHarmonics</code>	Number of harmonics displayed in the spectrum analysis window.	5
<code>HSSelectionToMarker</code>	If equal to 1, moves the insertion point to a marker newly created by clicking into the markers list.	1
<code>HSWindowOffset</code>	Default distance (in pixels) between the main document window and the left edge of the screen.	20
<code>HSNameTracks</code>	If equal to 1, automatically adds a “Name” and a “Track” tag to files created using the Split according to markers function.	0
<code>HSCreateIndexPoints</code>	If equal to 1, <code>index points</code> are created at markers not used to delimit tracks when burning a CD. For example, if black markers are used to delimit tracks, then all markers that are <i>not</i> black will be interpreted as index points. Note: Not all CD players support index points.	0

11 Sound analysis tools

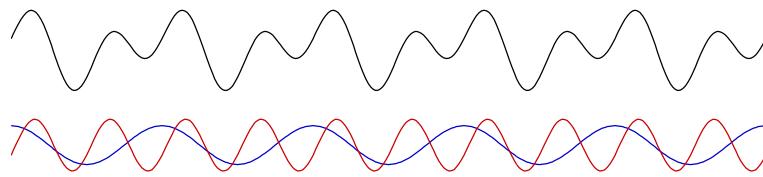
AmadeusPro comes with a number of high quality sound analysis tools. They fall into three categories: graphical analysis tools (sonogram and spectrum analysis), non-graphical tools (waveform statistics and find maximum), and real-time analysis tools (multi-band RTA, real-time spectrum, oscilloscope).

Most real-time analysis tools can be used in full-screen mode. To enter full-screen mode, choose **Full Screen** in the **View** menu or type command-L. To exit full-screen mode, type any key except for the space bar. The space bar instead pauses the tool.

Note that full-screen mode is not available for sound editing.

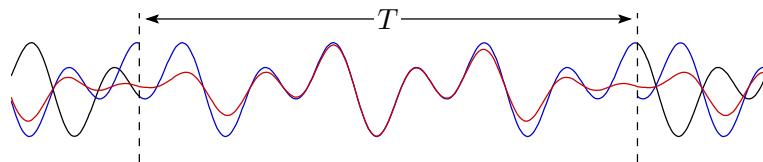
11.1 Spectrum

This is the most basic and common graphical analysis tool. It relies on the fact, known since the work of the French mathematician **Joseph Fourier** (1768–1830), that any periodic signal of a given frequency can be decomposed into a sum of ‘pure’ (or sinusoidal) signals whose frequencies are multiples of the frequency of the original signal. For example, the following picture shows the signal from page 5, as well as the two sinusoidal signals (blue and red lines) into which it can be decomposed.



The **Spectrum** tool takes a piece of signal of a certain length T (the length in question is equal to twice the number, in samples, selected in the **Size** field of the dialog that comes up when you choose **Spectrum**) and decomposes it into a superposition of sine waves of periods T , $T/2$, $T/3$, etc as explained above. It then represents the amplitudes of these pure sine waves as a function of their frequencies.

The location of the piece of signal in question relative to the current selection can be chosen in the **Spectrum location** field. At this stage, you may wonder about the fact that the period T chosen for the spectrum analysis is just a fixed number chosen by the user (like 4096 samples for example), which does not necessarily correspond to the period of the signal that is being analyzed. Furthermore, it may well happen that the signal in question is just not periodic at all. What **AmadeusPro** (and just about any other analysis tool) does is to consider the *periodic continuation* of the signal. In other words, it just pretends that the signal is periodic of the given period, even though this is probably not the case. Let us see what this means for our favourite signal (still the one from page 5). The following figure shows that signal in black, and it has an interval marked as corresponding to the period T which has nothing to do with the true period of the signal.



The blue line then shows to the periodic continuation of the signal according to the marked interval. Note that this line is discontinuous, *i.e.* it has a ‘jump’ at the endpoints of the marked interval. It is intuitively clear that this discontinuity is not a desirable feature. One way of getting rid of it is to multiply the signal by a *windowing function* which is close to one inside the marked interval and goes to zero at its endpoints. The red line in the above figure shows the periodic continuation of the ‘windowed’ signal. It does no longer present a discontinuity but of course it does no longer perfectly match the original signal inside the marked interval. Such a windowing function can be selected in the **Windowing function** field when you make a spectrum analysis.

A list of the most popular windowing functions can be found at this [Wikipedia page](#).

The spectrum analysis window has several features that facilitate its reading. The **Current location** text field in the toolbar shows the frequency / intensity corresponding to the current location of the mouse. Intensities are given in decibels. The **Nearest note** text field shows the note (on a standard **tempered scale** with A4 at 440Hz) which is nearest to the frequency corresponding to the current location of the mouse. The exact frequency corresponding to that note is shown in brackets.

In order to facilitate the reading, a thin green line is drawn at the current location of the mouse. Furthermore, the **harmonics** (*i.e.* multiples) of the corresponding frequency are also shown as green lines. The reason why harmonics are important is because a note played on a musical instrument does not consist of a single frequency but of a **fundamental frequency** and all its multiples, or harmonics. The ratios between the

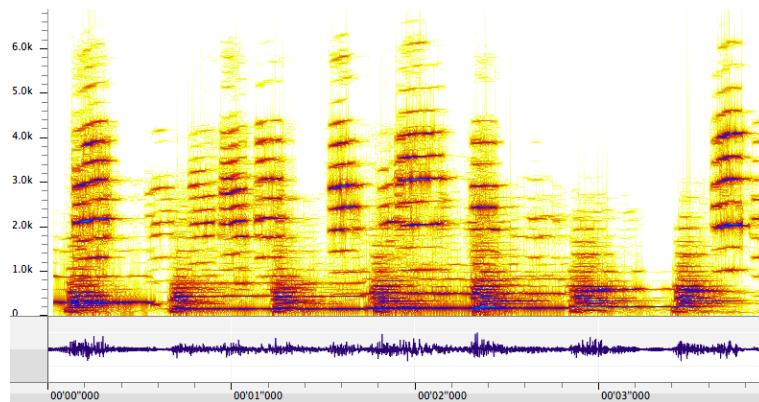
intensities of the harmonics determine the ‘color’ of the sound and allow us to distinguish between different musical instruments. The number of harmonics that are displayed is 5 by default, but can be changed by a hidden preference, see Section 10.

The scale of the display can be changed by clicking on the **Settings** icon in the toolbar.

A spectrum analysis can be exported as a text file for processing in another program by choosing **Save A Copy As...** in the **File** menu. This creates a file with three columns. The first column contains the frequency of a given component, the second and third contain its amplitude in complex notation. (So the intensity of a component is the sum of the squares of the real and the imaginary part.) If the spectrum was created by choosing **Average over selection**, then all the imaginary parts are 0.

11.2 Sonogram

A sonogram (also called **spectrogram**) is a two-dimensional graphical representation of a sound. An example of a sonogram is shown below:

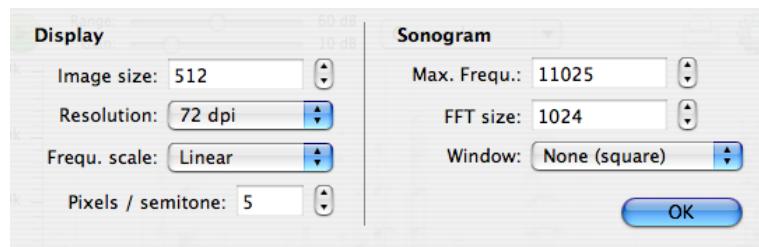


The bottom part of the window contains a waveform representation of the sound, just like in the main document window. The top part represents the frequency as a function of time, with intensities shown with a color scale. With the scale chosen for this picture, the white and yellow parts are of low intensity whereas the red and blue parts are of high intensity.

Basically, each vertical ‘slice’ of a sonogram corresponds to a spectrum analysis as described in the previous section, except that intensities are displayed with a color scale. When hovering over a sonogram, the frequency (and nearest note in brackets) corresponding to the current location of the mouse is displayed in the lower left corner of the window. A sonogram window can be navigated in pretty much the same way as a sound document window. Most of the same key shortcuts work in both windows. The **Color palette** menu of the toolbar allows you to choose between different color scales, as well as to edit them.

The **Range** and **Gain** sliders in the toolbar allow to adjust the color scale. Its effect is probably best explained by an example. Let’s say that one chooses the **Grayscale** color palette which draws areas of low intensity in white and areas of high intensity in black. Then, if **Range** is set to 60 and **Gain** is set to 15, every intensity above -15dB will be drawn in black and every intensity below -75dB ($75 = 60 + 10$) is drawn in white. Intensities falling between these two extreme values will be drawn in shades of gray.

Clicking on the **Settings** icon in the toolbar shows the following dialog:



The **Image size** field allows you to choose the height of the sonogram in pixels and the **Resolution** field allows you to choose its resolution. This latter setting is only used when printing or when exporting the sonogram to an image file. The **Frequ. scale** field allows you to switch between a linear and a logarithmic scale for frequencies. In a linear scale, the distance between successive harmonics of a given frequency is constant, whereas in a logarithmic scale it is the distance between successive **octaves** which is constant. The **Pixels / semitone** setting is only relevant if the frequency scale is chosen to be logarithmic. It allows you to set the distance (in pixels) between two successive **semitones** on the sonogram. Note that there are 12 semitones in an octave, so that if this value is for example set to 10, then two successive octaves will be 120 pixels apart.

The **Max. Frequ.** setting allows you to change the maximum frequency displayed in the sonogram. This is always a **subharmonic** of the **sampling rate**. The **FFT size** and **Window** settings have exactly the same meanings as the **Size** and **Window** settings for the **Spectrum** function. Note that the **FFT size** value does not necessarily need to match the **Image size** value. Larger values tend to give a ‘crisper’ picture, whereas lower values will give a more blurred picture. It is usually a good idea to use some windowing in order to get a nice sonogram representation of a sound.

Sonograms can be exported as pictures in various formats by choosing **Save A Copy As...** in the **File** menu. They can also be saved in a format proper to **AmadeusPro** by choosing **Save** in the **File** menu. Note that this format is not compatible with sonogram files created by Amadeus II.

11.3 Waveform Statistics

As its name suggests, this function allows you to collect various statistics on the selected sound. The various **RMS Power** values are calculated by computing the RMS power over intervals of length **RMS window**. The **DC Offset** value represents the average value of the waveform. It is usually desirable that this value is low. If it is larger than about 0.01 over an interval of more than one second, you may want to use the **Suppress DC Offset** function in the **Effects ⇒ Denoising** menu.

Clipped samples are samples that take the maximal admissible value. The presence of clipping indicates that the volume of the sound is too high. This usually results in a ‘saturated’ sound that is very unpleasant. Unfortunately, it is not possible to significantly improve the quality of the sound once clipping has occurred. If you find that a recorded sound contains clipped samples, it is advisable to perform the recording again with the input gain set to a lower value.

The **Average RMS power** is a useful indication of how ‘loud’ a song sounds on average. If you have two songs with one of them showing an average RMS power of –20dB and the other one showing a value of –30dB, the second song will sound significantly quieter than the first one. In order to bring it to roughly the same level, you can amplify it by 10dB. See also Section 5.3.5 for some tips on how to normalize sounds to bring them approximately to the same level.

Hitting command-C while the waveform statistics are displayed copies them into the clipboard, so that they can be reused for example in a spreadsheet program like OpenOffice.

11.4 Find Maximum / Find Next Clipped Sample

This function allows you to find the largest sample in the current selection. If the selection is empty, it will find the next clipped sample, starting from the location of the insertion point. Once the search is complete, the insertion point is moved to the corresponding location.

11.5 Multiband RTA

This is a real-time analysis tool that displays the frequency content of the sound currently being recorded. The full frequency range (20Hz – 22kHz) is subdivided into 10, 20, or 30 bands according to a logarithmic scale and the intensities of each band are displayed. The values are averaged over time in order to avoid fluctuations which are too fast. The duration of this averaging can be chosen with the **Speed** menu.

11.6 Real-Time Spectrum

This is essentially the same as the **Spectrum** function, except that it works in real-time on the selected sound input.

11.7 Oscilloscope

This tool allows you to display in real-time the waveform of the sound signal on the selected sound input.

11.8 Real-Time Sonogram

This tool works exactly like the **Sonogram** function, but it takes its input in real-time from the selected sound input. The speed at which the display moves can be adjusted by clicking on the **Settings** item in the toolbar.

11.9 Real-Time Waveform

This tool displays the amplitude of the signal of the selected sound input in real time. It is convenient for monitoring the sound intensity of a recorded signal, in order to make sure that no clipping is occurring. The speed can be adjusted as either 10, 20 or 40 frames / second (slow, medium, fast).

12 Miscellaneous functions

12.1 Memory management

The **General** preference pane contains the somewhat esoteric **Maximal RAM usage** setting. It allows you to determine how much sound data **AmadeusPro** puts into memory before it starts using temporary files instead. The advantage of using memory instead of temporary files is that it is much faster to access and will therefore result in a much ‘snappier’ experience. This however is only true if a sufficient amount of

memory is indeed available. If not, the system will use virtual memory instead, which is much less efficient than using temporary files. As a general rule, keep the maximal RAM usage below about one quarter of the total amount of RAM installed in your computer.

12.2 Version checking

The default behaviour of **AmadeusPro** is to regularly check for the availability of new versions of the program if it detects an active internet connection. This check happens weekly, but you can either change this frequency or disable it altogether in the **Updates** section of the preferences window. No personal information is transmitted during these checks, they consist in simply accessing [this file](#). If you don't trust what software companies tell you (and you would probably be wise not to take everything at face value) you are of course welcome to check that this is true by using `tcpdump` or your favourite TCP/IP packet sniffer.

If a new version is detected, **AmadeusPro** can download the update and install it without any user intervention. The default behaviour is to ask the user for permission before starting the download, but if the **Download updates automatically** checkbox is checked, downloads will start automatically without warning. (It will however present a progress window through which the download can be cancelled.) Installing the download then requires **AmadeusPro** to quit and relaunch itself. If your user account does not have sufficient privileges to erase the old version of **AmadeusPro** and replace it by the new version, you will be asked for an administrator password.

Whenever **AmadeusPro** checks for the presence of a new version of the application, it also checks for new versions of the manual. If a new version is detected, you will be given the opportunity to install it onto your system. The manuals are kept in the **Application Support/Amadeus Pro** subfolder of your **Library** folder. The reason for keeping the manual separately from the application is to save bandwidth. The manual currently exists in four languages, but an individual user is unlikely to need more than one of them. Furthermore, the manual is usually not updated between minor releases of the application.

12.3 Updates of the manual

Whenever **AmadeusPro** checks for a new version of the program, it also checks whether an updated version of this user's manual is available. If there is, you will be notified and asked whether you want to download it. If you decline, you will be asked again the next time that **AmadeusPro** checks on it.

12.4 About the licensing system

12.4.1 Purchasing a license

Payments for **AmadeusPro** are handled by the payment processor [Kagi](#) and all network connections are secured with the latest encryption technology. There are two ways of purchasing a license for **AmadeusPro**:

1. **Online purchase.** The [online shop](https://order.kagi.com/?L851) at <https://order.kagi.com/?L851> provides the most flexible way of purchasing an **AmadeusPro** license. Besides accepting all major **credit cards**, it also accepts various alternative payment forms, like **cash**, **checks**, and **payment orders**. The online shop is also the only way of purchasing **site licenses** and **multiple licenses**.

Your license will be included in an email with the subject **Thanks for your purchase** sent to you by Kagi. Please make sure to set up your spam filter in such a way that this message does not get

discarded! Once you have a valid license, open **AmadeusPro** and click on **Register Now...** when presented with the initial ‘This copy of Amadeus is unregistered!’ dialog. Make sure to enter both your name and your license key exactly as they appear in the ‘Thanks for your purchase’ email. Since licenses are tied to the customer’s name, using a different name will result in the license being rejected.

2. **In-program purchase.** An other way to purchase a license for **AmadeusPro** is directly from within the application. This has the advantage of streamlining the registration procedure. Click on **Register Now...** when presented with the initial ‘This copy of Amadeus is unregistered!’ dialog. Then, click on the **Buy Now...** button. This will bring up a dialog asking for your registration and credit card details. **An active internet connection is required for this!**

After the license has been successfully purchased, it will automatically be entered in the registration dialog by **AmadeusPro**. A copy will also be sent to you in an email with the subject ‘Thanks for your purchase’.

Important Note: There will be **no physical shipment** taking place in any case. The latest version of **AmadeusPro** can be downloaded on the [HairerSoft](#) website. The address that you enter during the registration process will only be used by the fraud prevention mechanism to verify that your credit card is valid.

12.4.2 License management

The **Amadeus Pro** menu gives you access to a license manager that shows which licenses for **AmadeusPro** are installed on your system:

Licenses installed for the software Amadeus Pro:				
Name	Type	Date	Expires	
HairerSoft	Site License	07/24/2007	Never	
Martin Hairer	Single User	07/24/2007	Never	

[Delete license](#) [Add license...](#)

For example, the above system has one single-user license for ‘Martin Hairer’ and one site license for ‘HairerSoft’. Site licenses are accessible to all users on the system, whereas single-user licenses are only accessible to one user.

The license manager allows you to add new licenses and to remove existing licenses (for example, you should remove your license before giving your computer away).

The most important part of the [single-user license agreement](#) is:

Grant of License. HairerSoft hereby grants to you, an individual purchasing a single-user license, the right to install and activate **AmadeusPro** on up to three separate computers for your personal use. Sharing this Software with other individuals is in violation of this license.

In a nutshell, one single-user license allows you to use **AmadeusPro** on your laptop, your home computer and your work computer. However, if your spouse wishes to use **AmadeusPro** as well, two licenses are required. The precise terms of the single-user license can be found at [this link](#).

Similarly, the most important part of the site license agreement is:

Grant of License. HairerSoft hereby grants to you, the organization / school / company / university purchasing a site license, the right to install and activate **AmadeusPro** on an illimited number of computers belonging to you, provided they are located within a radius of 10km of each other. It can be used on these computers by any individual affiliated to you. Sharing this Software with individuals not directly affiliated to you is in violation of this license.

So, if you are an arts school purchasing a site license, your staff and students can use **AmadeusPro** on the university-owned computers on campus. Your are however not allowed to give away copies of the license for the staff / student's use on their personal computers. Also, if you have a second campus in another town, you will need a second site license. The precise terms of the site license can be found at [this link](#).

12.4.3 Upgrading from Amadeus II

Registered users of Amadeus II are entitled to purchase an upgrade license for **AmadeusPro** at a discounted price (\$25 instead of \$40). If you purchase a license from within **AmadeusPro** by clicking on the **Purchase now...** button in the registration dialog and a valid Amadeus II license is detected on your system, you will automatically be led to the purchase of an upgrade license. Otherwise, you will be led to the purchase of a regular license. If this is not what you want, please purchase the requested type of license in the [online shop](#).

An upgrade license is only valid together with an Amadeus II license. If **AmadeusPro** cannot detect the presence of a valid Amadeus II license on your system, it will ask you to enter your Amadeus II license when registering.

12.4.4 Lost licenses

First of all, **please keep your license in a safe place**. There is no guarantee that a lost license can be replaced. If it happens nevertheless that your license is lost, you can request a copy of it by sending an email to LostLicense@HairerSoft.com. Please include in your message both your name and the email under which you originally registered **AmadeusPro**. In order to avoid fraud, replacement licenses will always be sent to the email that you originally used to register **AmadeusPro**.

If you change your email address, please send an email **from your old account** to ChangeEmail@HairerSoft.com. This email should contain your new address in the body. A confirmation email will be sent to both your old and your new address confirming the change.

The procedures described in this section are not fully automated and it may therefore take several working days before you get a reply.

13 How to's

13.1 How to work with your iTunes music library

There is nothing easier than editing a sound from your iTunes music library:

- Right-click (or control-click) on the song in iTunes and select **Show in Finder** from the contextual menu.
- Drag the file from the Finder onto the **AmadeusPro** icon in the dock.
- After you are done editing the sound, just select **Save** in the **File** menu to save the sound back to the same place.

In order to move a file edited with **AmadeusPro** to iTunes, the easiest way is to either use the dedicated **Export to iTunes Library** function from the **File** menu or to drag the file from the Finder onto the iTunes library.

Warning: Apple does not allow third parties to read protected AAC files. Songs purchased on the iTunes music store can therefore **not** be edited with **AmadeusPro**, unless they were purchased from **iTunes Plus**. The easiest workaround to this situation is to burn the song to a CD with iTunes and then to open it from the CD with **AmadeusPro**.

13.2 How to convert a sound from one file format to another

A very common operation is to convert a sound from one format to another one, for example to encode a song from a CD to either Mp3 or M4a or to create FLAC files for archiving. Just open the sound into **AmadeusPro** by using the **Open** command from the **File** menu or by dragging the icon of the sound you wish to convert from the Finder onto the **AmadeusPro** icon in the dock.

Then select **Save As...** from the **File** menu and select the desired destination format in the pop-up menu located at the bottom of that dialog. You may optionally wish to tweak the settings associated to that particular format by clicking on the **Settings...** button. See Section 3 for a description of the various formats and the settings that go with them.

13.3 How to create a CD from a tape or vinyl recording

It is possible to achieve this without ever leaving **AmadeusPro** and without installing any additional software by following these steps:

- Record the tape or vinyl to a new document, as explained in Section 4.1. Make sure that the recording is performed in stereo with a sampling rate of 44.1kHz.
- Optionally remove any background hiss or cracks as explained in Section 8.
- Place markers at the junctions between songs. In most cases, this can be done automatically with the **Generate Markers...** function in the **Selection** menu. If this doesn't work (for example in a live recording), you can place them manually. Markers mark the beginning of a track, so that in most instances you would want to place a marker at the very beginning of your file.
- Choose **Burn to CD...** from the **Sound** menu. The top part of this dialog allows you to choose which markers to take into account for splitting the document into tracks. If you wish to create a CD with gapless playback (when there are smooth transitions between songs), you should choose 0 as the gap size. Otherwise, 2 seconds is a standard value. You can create a CD with CD-TEXT information which will then display the name of the currently played song and the title of the album on certain CD players. The title of a song will be given by the name of the marker which marks the start of the song.

You should therefore make sure that there is a marker located at the very beginning of the document as well.

If the document contains songs by different performers, it is possible to store this information within CD-TEXT as well. You should choose either 'Performer:Title' or 'Title (Performer)' in the 'Markers' menu and name your markers accordingly. For example, if your document contains the song 'Breathe' performed by Pink Floyd, you should name the corresponding marker either 'Pink Floyd: Breathe' or 'Breathe (Pink Floyd)' depending on which naming convention you choose.

You can also write a media catalog number to the Audio CD. If you don't know what this is, then you will certainly not need this and you can safely ignore this option.

- After clicking **OK**, the standard system CD burning dialog will give you the choice of which CD burner to use and at which speed to burn the CD. Insert a blank disk and hit **OK** again. You are done.

Important: Before you can burn an **AmadeusPro** document to CD, you need to make sure that it consists of a single stereo track and that its quality is 16 Bit / 44.1 kHz. If it consists of more than one track, please use the **Flatten Document** function from the **Sound** menu in order to combine the tracks into one. If the quality is different from 16 Bit / 44.1 kHz, use the **Characteristics...** function from the **Sound** menu to adjust it.

13.4 How to record from the output of another program

Sometimes, you may wish to record the output of another program (for example **iTunes** or **Safari**). One possible way of achieving this is to connect the sound output of your computer (the headphone port) back into the line in input. This is however not very elegant and results in a slight loss of quality due to the two analog / digital conversions that take place.

Solving this problem in a more elegant way is not possible without the help of some additional software. One possible solution is to use the freeware **Jack** in conjunction with **AmadeusPro**. After downloading Jack from <http://www.jackosx.com/> and installing it, you can record sound from the output of, say iTunes, by following the steps below. This procedure has been tested with iTunes 7 and JackPilot 0.74; other versions may behave in a slightly different way.

Important: Do not use this procedure to import a sound from your iTunes library into **AmadeusPro**, unless it is a protected AAC file. For unprotected files, use the procedure described in Section 13.1.

1. Make sure that **AmadeusPro** and iTunes are both closed.
2. Open the JackPilot program.
3. Go to the JackPilot preferences and choose **Built-In Output** (or whatever your preferred output device is) in **Interface**. Also make sure that the option **Auto-Connect with physical ports** is unchecked.
4. Click on **Save** in the preferences window and then start the Jack daemon by clicking on **Start** in the **JackPilot** window.
5. Open the **Sound** panel in the system preferences and choose **Jack Router** as the default output.
6. Start iTunes and **AmadeusPro**.

7. Open the **AmadeusPro Sound** preferences and choose your preferred output device (for example **Internal Speakers**) in the **Output** panel and **Jack Router** in the **Input** panel.
8. Choose **Record to New File...** in the **Sound** menu (or option-click on the record button).
9. Bring the JackPilot application to the front and click on **Routing**.
10. Click on the little arrow next to iTunes in the **Send Ports** column in order to expand it and select the first port (labelled **out1**).
11. Click on the little arrow next to **AmadeusPro** in the **Receive Ports** column in order to expand it and double-click on the first port (labelled **in1**). It should turn red and an entry should appear in the **Connections** column.
12. Repeat steps 10 and 11 for the second port (**out2** and **in2**).
13. Start your recording in **AmadeusPro**. Enable the **Playthrough** function in the recording window in order to hear the sound you are going to play.
14. Start playing a song or a radio station in iTunes.

Note that it is also possible for example to create a virtual four-channel device with Jack. In this way, you can for example route the output of iTunes into the first two channels and your microphone into the last two channels. It is then possible to record all four channels simultaneously with **AmadeusPro** in order to create a document with a vocal and a music track.

13.5 How to add a voice track to an existing music track

Starting from version 1.2 of **AmadeusPro** it is very easy to add a voice track to an existing music track in a perfectly synchronised way:

1. Open the music track from your hard disk.
2. Set **AmadeusPro** up to record from your microphone (see also Section 4.1).
3. Choose **Record to New Track...** in the **Sound** menu.
4. If recording does not start immediately, click on the red recording button. This will automatically start the playback of the existing music track.
5. Give your best to create a beautiful voice track!
6. Once the music stopped playing, click on **Stop**. This should add a new track to the existing document.
7. Play back your recording and adjust the relative volumes of the two tracks by playing with the **Volume** gliders.
8. If you want to mix the music and the voice into one single track (this is optional, but recommended before saving to an AIFF or Mp3 file), choose **Flatten Document** in the **Sound** menu.

14 Troubleshooting

The most common causes for problems concern the recording / playback functions since these are the ones where it is easiest to change the settings by mistake in such a way that the resulting behavior appears strange...

14.1 I can't hear anything when playing back

1. Check that you have selected the correct output device in the **Output** tab of the **Sound** preference panel. (We are talking about **AmadeusPro**'s preference panel here, not the system one! The settings there override those in the system preferences.)
2. Make sure that the **Play** checkbox is checked for the track that you are playing back and that its volume is sufficiently high.
3. Reboot your computer. It happens (rarely, but still) that the audio driver gets into a strange state and refuses to work properly.
4. If you are using external hardware for playback, check that it is connected properly to your computer. Can you get iTunes to play back through it?
5. It might be that you are not directing sound to the correct hardware channels. Change the channel association in the **Advanced Settings** dialog accessible from the **Output** pane of the **Sound** preferences panel.

14.2 I don't hear what I am recording

1. Make sure that the **Playthrough** checkbox is ticked in the recording window.
2. Follow the steps above to make sure that playback works.
3. Are the **vu-meters** moving? If not, follow the steps below to make sure that recording works.

14.3 I can't record anything or the recording is not what I expect

1. Check that you have selected the correct input device in the **Input** pane of the **Sound** preference panel. (We are still talking about **AmadeusPro**'s preference panel here, not the system one.)
2. Make sure that the **Gain** sliders are sufficiently far to the right.
3. If you are using external hardware for recording, check that it is connected properly to your computer. Do the vu-meters move if you select that device in the **Input** tab of the **Sound** system preferences panel?
4. If you are using a microphone, make sure that it either includes a preamplifier or that it is connected to a 'microphone' input. Most models of Macintosh have a 'line-in' input instead which does *not* include a preamplifier. One of the cheapest preamplifiers around is the iMic from Griffin's. (Make sure that the little black switch on it is in the correct location.)
5. If you are using the **Record to new File...** function, make sure that you have selected the desired quality (including the channel configuration) in the **Quality** tab of the recording window.

6. If you are using the **Record** or the **Record to new Track** function, it will create a recording with the same characteristics of the currently selected audio track. Make sure that this is what you want.
7. As above, it might be that you are not recording sound from the correct hardware channels. Change the channel association in the **Advanced Settings** dialog accessible from the **Input** pane of the **Sound** preferences panel or equivalently from the **Input** pane of the **Record to new file** window.
8. Reboot your computer, it might be that the audio drivers got into a ‘bad’ state.

14.4 Frequently Asked Questions

A list of frequently asked questions is maintained at the following address:

<http://www.hairersoft.com/AmadeusPro/AmadeusProFAQ.html>

Furthermore, tips and discussions on using **AmadeusPro** can be found on the user’s forum at

<http://www.hairersoft.com/forum/>

The remainder of this section contains the state at the time of “printing” of the FAQ:

14.4.1 What is the upgrade policy for registered users of Amadeus II?

Registered users of Amadeus II can purchase **AmadeusPro** for US\$25 instead of the usual price of US\$40. They will be issued with that special registration code that only works together with a valid registration code for Amadeus II.

14.4.2 What are the differences between Amadeus II and Amadeus Pro?

The main differences are:

- **AmadeusPro** is a Universal Binary, i.e. it runs at native speed on Intel Macs. However, it requires MacOS 10.4 or later to run.
- **AmadeusPro** has a completely redesigned MacOS X - like Cocoa user interface.
- **AmadeusPro** allows multi-track editing and recording.
- **AmadeusPro** includes a batch processor.
- **AmadeusPro** can burn Audio CD’s directly from within the program.

There are many more small differences between the two and the best is to just try them out!

14.4.3 Why are CD track names all wrong on CDs burned by Amadeus Pro?

They only appear so in iTunes. The reason is that instead of correctly reading the CD-TEXT information contained on the CD (which is what a CD player does), iTunes tries to be smart and queries an online database (Gracenote) that tries to recognise the CD based on the lengths of the tracks. Chances are that your home-made CD will be matched with some completely unrelated commercial CD that just happens to have the same (or similar) track lengths, thus the resulting garbage. Apart from lobbying Apple to fix that bug (feature?) of iTunes, there's not much that one can do. To be fair, one possibility would be to burn the CD as a data CD instead of an Audio CD, but then most CD players wouldn't play it.

As you can guess from the explanation, this problem is not specific to **AmadeusPro** at all. It is just the same if you burn the CD with iTunes or any other other CD burning software.

14.4.4 Could you add a key shortcut for function XYZ?

Go to the “Keyboard and Mouse” system preferences panel. On MacOS 10.3 and later, it allows you to assign key shortcuts to arbitrary menu items.

14.4.5 Isn't it possible to undo actions in the repair center?

Yes it is! The reason why the **Undo** menu item can be grayed out is just because it applies to the frontmost window, which might be the repair center instead of the sound window. So, just select the sound window and you can undo as usual. You can also use the handy ‘z’ key shortcut (no command, just ‘z’) to undo and ‘Z’ to redo.

14.4.6 Function XYZ from Amadeus II is missing in Amadeus Pro. Are you going to add it?

The **Filter...** function from the effects menu is probably not going to be added back, since its functionality is duplicated by the multiband equalizers. The 3D-spectrum analyzing tool will probably not be added back either, since it didn't seem to be used by sufficiently many users. I believe that all other functions that were present in Amadeus II have been replaced by equivalent (or more powerful) functions in **AmadeusPro**. If you disagree, please let me know.

14.4.7 The sliders for the record levels are greyed out on my computer and the input level is inappropriate, why?

These sliders directly control the hardware. Therefore, they are only active if your hardware actually does provide gain control and the driver of your hardware implements it correctly. If your particular hardware doesn't support it, there's nothing that one can do about it. You should be careful to adjust the level of the signal to the correct level before it reaches your hardware (or using the controls of the hardware if it has some). Also, make sure that you are actually using the correct sound input device. You can check whether a particular device supports gain adjustment with the “Audio MIDI Setup” application in your Applications/Utilities folder (select the device and check whether the input sliders there are greyed out).

Note that the iMic has a little black switch that allows you to switch between microphone level input and line-in level input. Make sure that this switch is in the appropriate position.

14.4.8 Do you accept PayPal payments?

Yes. Click [here](#) and follow the on-screen instructions. You will be issued with a valid license as soon your payment has cleared, which should be instantaneous unless you are using an e-Cheque.

14.4.9 Is there an academic / nonprofit organisations discount?

I am sorry, but there isn't any discount. The price of **AmadeusPro** has been fixed at the absolute minimum that is still viable.

14.4.10 Amadeus attempts to make an internet connection at startup! What the heck?

Occasionally, Amadeus makes an http query to see whether a new version is available. What happens is exactly the same as what happens when you click on [this link](#) (i.e. it just retrieves a piece of text). If you are concerned about this or choose not to believe what I say, you can disable version checking in the "Updates" section of the **AmadeusPro** preferences.

14.4.11 I can not open m4p files purchased on the Music Store with Amadeus, why?

With QuickTime 6.5.1 (the update coming with iTunes 4.5), Apple decided without prior notice to forbid third-party applications the access to m4p files. One workaround is to play the files in iTunes and to use [Jack](#) to record the output from iTunes. Another workaround is to burn the file on an Audio CD and to open it from there.

14.4.12 What is an RIAA curve?

In general, vinyl recordings are engraved with reduced bass levels and increased treble levels. In 1954, the [RIAA](#) (Recording Industry Association of America) introduced a standard filter that was used on nearly all phonograph recordings since then. The filter "RIAA" available from "Filters" in the "Effect" menu allows to compensate this filter in order to reproduce vinyl recordings in the correct way.

15 Acknowledgment to third party software

AmadeusPro relies on a number of third-party open source software packages. Its functionality would be nowhere near what it is today without these fabulous gifts to the community. They are listed in this section, together with the corresponding internet links and licensing information.

15.1 Ogg Vorbis

The libraries VorbisLib and VorbisFileLib are licensed under a BSD-type license. Its terms are:

Copyright (c) 2002, [Xiph.org](http://www.xiph.org) Foundation

Redistribution and use in source and binary forms, with or without modification, are permitted provided that the following conditions are met:

- Redistributions of source code must retain the above copyright notice, this list of conditions and the following disclaimer.
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The libraries OggLib and VorbisEncLib are licensed under the LGPL (GNU Library General Public License). For a copy of the LGPL, see the link

<http://www.HairerSoft.com/LGPL.txt> .

All the source code used to build these libraries can be obtained at

<http://www.xiph.org/ogg/vorbis/download.html> .

15.2 LAME

LAME is the package used to provide Mp3 encoding in **AmadeusPro**. LAME is licensed under the LGPL (GNU Library General Public License). For a copy of the LGPL, go to the link

<http://www.HairerSoft.com/LGPL.txt> .

The LAME source code can be obtained at

<http://www.mp3dev.org/mp3/> .

The copy of lame used in **AmadeusPro** uses a patch allowing to add a specified number of null bytes at the beginning of a file. This is so that **AmadeusPro** can then write the metadata to the file by itself. This patch can be obtained at

<http://www.hairersoft.com/Downloads/lamepatch> .

15.3 mpg123

mpg123 is the package used to provide Mp2 and Mp3 decoding in **AmadeusPro**. mpg123 is licensed under the LGPL (GNU Library General Public License). For a copy of the LGPL, see the link

<http://www.HairerSoft.com/LGPL.txt> .

The mpg123 source code can be obtained at

<http://www.mpg123.de/> .

15.4 FLAC

FLAC is licensed under the LGPL (GNU Library General Public License). For a copy of the LGPL, see the link

<http://www.HairerSoft.com/LGPL.txt> .

The FLAC source code can be obtained at

<http://flac.sourceforge.net/download.html> .

15.5 AtomicParsley

AtomicParsley is the software used to read and write metadata from and to Mp4 files. AtomicParsley is licensed under the LGPL (GNU Library General Public License). For a copy of the LGPL, see the link

<http://www.HairerSoft.com/LGPL.txt> .

The AtomicParsley source code can be obtained at

<http://atomicparsley.sourceforge.net/> .

15.6 MTCoreAudio

The MTCoreAudio framework is licensed under the following terms:

MTCoreAudio.framework is Copyright (c) 2004 by Michael C. Thornburgh zenomt@armory.com. All rights reserved.

MTCoreAudio.framework and its source code may be used, modified, incorporated, etceterated for free by anyone for any purpose. I request, but do not require, that credit for any use of the code is given. I request, but do not require, to be notified of the incorporation of this code into other works.

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PROFITS; OR BUSINESS INTERRUPTION) HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY, WHETHER IN CONTRACT, STRICT LIABILITY, OR TORT (INCLUDING NEGLIGENCE OR OTHERWISE) ARISING IN ANY WAY OUT OF THE USE OF THIS SOFTWARE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGE.

The MTCoreAudio source code can be obtained at

<http://aldebaran.armory.com/zenomt/macosx/MTCoreAudio/>.

15.7 AMPreferencePane

AMPreferencePane is free software available from

<http://www.harmless.de/cocoa.php>.

It is distributed under the [BSD license](#).

Copyright (c) 2007, Andreas Mayer (andreas@harmless.de)

15.8 ILCrashReporter

ILCrashReporter is free software available from

<http://www.infinite-loop.dk/developer/>.

It is Copyright (c) 2004 Claus Broch, Infinite Loop. All rights reserved.

The version used by **AmadeusPro** has been modified to fix a number of minor glitches. The source code for the modified version can be found [here](#).

16 Bug reporting

Even though I make every possible effort to track down bugs before a new version of **AmadeusPro** is released, the complexity of the project makes it unavoidable that some of them slip through. If you believe that you have found a bug in **AmadeusPro**, please report it to Martin@HailerSoft.com and I will make every attempt to correct it. However, please make sure to respect the following guidelines that will make things so much easier for both of us since I will be able to provide you with a more useful reply and there is a much higher chance that I will actually be able to fix things:

- Make sure that you are using the latest version of **AmadeusPro**. It may well be that the bug has already been fixed! You can check the number of the latest version (and of course also download it) at <http://www.hairersoft.com/AmadeusPro/>. Please do however confirm in your email that you are indeed using the latest version of **AmadeusPro**, as otherwise this will be first question I'll ask you.
- Please check the [FAQ](#) and read this manual to make sure that your bug is not a well-known problem.
Example: Please don't report the fact that **AmadeusPro** cannot read protected AAC files!

- Describe the bug in a short but precise way and, if possible, include detailed step by step instructions on how to reproduce it. State precisely what the observed consequences of your manipulations are and what consequences you expected instead. Include as much relevant information as possible. This usually comprises at least the version of your operating system and the characteristics of the sound you were working with when the bug occurred. **Example:** If the bug concerns the recording functions, state precisely what hardware you are using for recording.
- If the bug leads to a crash of **AmadeusPro**, there should be a dialog appearing that gives you the possibility of sending a crash log back to HairerSoft. Please do so, this will help improving future versions. Adding information on what you were doing when the crash occurred greatly helps in tracking down the problem.

16.1 Disclaimer

Just to avoid getting into any sort of legal trouble, I have to mention that I am not responsible for any damage or loss of data that may (however unlikely this is) be caused by this or any other version of **AmadeusPro** to your computer, whether this damage is caused by incorrect operation or by a bug in the program.

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